

HAMILTON and Area Packet Network

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Hamilton and Area Packet Network
Bulletin number 2.

Well, it seems not so long ago, number 1 of the series was dropped in the mail and I settled down to a few days rest before starting on collecting goodies for this one.

This time we are going to begin a segment which I believe will be both interesting and instructive! We will select a couple of papers from the literature which contain useful information for the packet enthusiast. This issue contains a paper by da Silva and de Mercado of the DDC about packet speech and one by Abramson reflecting on the initial ALOMA system at the University of Hawaii.

Another standard column is the bibliography:

1. I have included some quick and dirty references to the current (< 5 mos) literature on networks, packet switching and protocols. These references are meant to fill in the gaps between updates to the main bibliography.
2. The bibliography on digital communications, specifically modulation techniques, is now available (for \$10.00 per copy). It contains over 400 references to modulation schemes, models, measurements etc.
3. The complete original bibliography (parts I and II) on packet, all 31 double sided pages of it, is now available for the (unfortunately increased) price of \$10.00.

The suspiciously round figure of \$10.00 covers the cost of duplication and mailing, and contributes a small sum to the group for further exploration as well as putting you on the mailing list.

In the circuitry department, a super simple TRAP debounce circuit by John (DUV) and his polite multiplexed modem (makes his packet wait till he finishes talking before it is sent). Connections to a TRS-80 are given by Glenn (DSP) and Max (DNM), and Robert's (EFD) 'information detector' is revealed. (Robert uses a Yeasu hand-held without accessing the squelch circuit and detects channel occupancy using audio techniques with the result that he can sneak packets thru between words on a busy channel!)

The software segment contains John's TRAP dumper routine and Glenn's TRS-80 routine to talk to the TNC.

Because there was some concern about standardization, a small segment on protocols and their role is presented along with a (proposed) standard for screen control based on the ANSI standard. An untested piece of code for the SD Sales VDR-8024 board (uses Z-80) is given as an example of how the decoding of escape sequences is accomplished.

I apologize for the bad quality on the copy of the simple 8273/S-100 bus interface schematic in the last blurb. If you really want to build one and want the schematic, send a SASE to 2391 Arnold Cres. and I will send you a better copy.

You will find a Bras sheet enclosed. If you know of anyone who would like to become a member and receive a complimentary copy of the current bulletin, have him/her fill out the form and return it to the Hamilton and Area Packet Network, c/o Stu Reel, 2391 Arnold Cres., Burlington, Ontario, L7P 4J2, Canada.

Protocols

First, a definition (related to computer technology): Protocols are common tools designed for controlling information transfer between computer systems. The components of a protocol can include rules, message formats, frequency of interactions, error control, anticipation and flow control, accounting or synchronization and addressing strategy.

One observation is important at this point! If the system were simple enough and all users of the system had identical equipment, there would be little use for many of the protocol components! All users would merely copy the common communication program and use it. But computer systems (in particular the systems in use on the network) are anything but homogeneous, thus the requirement to identify the differences among them and take all this into account when what is called the 'Reference Architecture' is designed.

A cassette recorder and a radio can be connected together because the manufacturers agreed to standardize on miniature phone plugs as the 'Visibility Points'.

of their systems. One doesn't need to know the inner workings of the whole device, just the socket pins: the peephole into the device.

Computer systems are more complicated than this example, but the requirements for standardization are not. Even within a single system, there are requirements for protocols between the 'layers' of software and hardware: protocols for communicating from the application program to the exec, from the exec to the drivers, from the drivers to the buss, from the buss to the interface and from the interface to the device (and possibly further). One thing to stress is that you should be able to change things within a given layer with minimal (preferably no) interference between layers (the 'interface' between the application program and the exec shouldn't change if the driver changes (within reason)).

Once we extend our view to the network, particularly as implemented by the VADCB board, the computer terminal systems are decoupled from the net at the TIP program. There still exists an interface reference in the RS-232 or parallel link, but it doesn't impact the net in the slightest! one is free to design his own TIP and computer system.

The protocol components of 'Transmission' (RF link) and 'Network' (hop to hop validation) are already dealt with on the VADCB board as well as some rudimentary 'Flow Control' (return REJECT when buffers are full) but the application level components have yet to be provided, because all the applications have not all been identified. At the application level there is, however, a requirement for at least two components: the 'Virtual Terminal' and 'File Transfer' protocols. (The File Transfer protocol will be dealt with in another issue of the bulletin.)

What is a Virtual Terminal?

The protocol required to allow all terminals on the net to talk to each other and issue a common control language is the Reference Architecture of the Virtual terminal. First we identify all the functions of all the terminals in use. Then we do one of two things: a) standardize on the functions and mechanism of one of the terminals and make all the others emulate it or b) select a new definition and make all emulate it.

The first approach suffers in that if the chosen device doesn't have some feature that another does, the

feature cannot be used at this protocol level: a higher level must be instituted to allow it's use.

The second approach is more general and will include the 'OR' of all pertinent features, but an implementation is required for all types of terminals.

What we will now propose is a standard for the Virtual Terminal. Please note all that was said about protocol rules, in particular, changes in one layer should not impact others. When you are designing software, continuously ask yourself 'If some other layer changes, how much code will I have to re-write?' and then decide on the trade off you are willing to make.

OK. The protocol for the Virtual Terminal is that selected by the American National Standards Institute. It is based on the 'Escape Sequence'.

The functioning of a terminal is composed of two aspects: communication and formatting. The keying and display of characters communicate the desired information, the arrangement of the characters on the screen (or page) is formatting. The Virtual Terminal displays information unless it has been prompted (by the Escape or some special Control character) to perform formatting. The next few pages describe the implementation of the ANSI standard on a MIM 100 terminal (Lanpar XT-100). A simple tree to examine escape sequences is then presented. This code is untested and should only be used as a guide to an implementation. It was written for a Z-80 chip, so the table look up will have to be replaced as well.

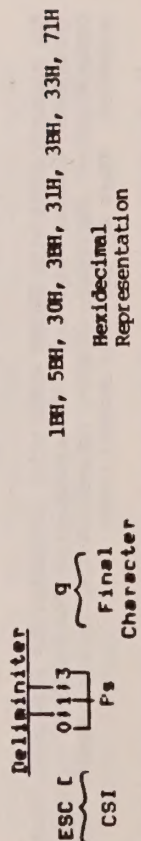
On the other side of the coin is the argument that protocol standardization can lead to unwieldy operations when conversions are done between systems. It is thus necessary to allow individual users pairs to select their own private protocols for efficiency purposes. The provision of general protocols will be necessary for the simple or once-thru operations when designing a custom protocol is not required. In the Hamilton net, some users are concerned about the virtual terminal problem and are working on it, while others are concerned with the applications and the protocols will develop out of the implementation.

This has been a brief and cursory statement about protocols. More information is available in 'A Tutorial on Protocols' by Pouzin and Zimmerman in Proc IEEE, vol 66, no.11, 1978, p1346-1370.

Examples: A) COMMAND: Move the active (cursor) position to the right 15 places.

ESC [15 C 1BH, 5BH, 3BH, 3BH, 35H, 43H
CSI PM Final Hexidecimal
Character Representation

B) COMMAND: Clear lamps 1-4 illuminate lamps 1 and 3.



In general, a control sequence with several selective parameters in functionally equivalent to several control sequences, each with one selective parameter. The same results as the control sequence of example B) above could be accomplished by:

ESC [0qESC [1qESC [3q
α: ESC [1,003q
α: ESC [qESC [1,3q

The following section describes each escape sequence valid for the MIME 100 in ANSI mode. A summary is presented in Section 2.9.1.

2.7 ANSI CONTROL SEQUENCES

CURSOR POSITION REPORT
ESC [Pn;PmR

This sequence reports the active (cursor) position of the MIME 100 to the host by means of two numeric parameters (Pn), the first specifying the line and the second specifying the column.

The numbering of lines depends on the state of the Origin Mode.

This CONTROL sequence is solicited by a Device Status Report sent from the host.

CURSOR BACKWARD
ESC [PnB

This sequence moves the active (cursor) position to the left as determined by the numeric parameter. I.e., if the value represented by the parameter value is N, the cursor is moved N spaces to the left; if zero or 1, the cursor is moved one space to the left. When attempting to move the cursor beyond the left margin, the cursor stops at the left margin.

CURSOR DOWN
ESC [PnB

This sequence moves the active (cursor) position downward without altering the column position. The distance moved is determined by the numeric parameter; i.e., if the parameter value is N, the cursor is moved N lines downward, if zero or 1, the cursor is moved one line downward. When attempting to move the cursor below the bottom margin, the cursor stops at the bottom margin.

CURSOR FORWARD
ESC [PnC

This sequence moves the active (cursor) position to the right as determined by the numeric parameter. I.e., if the parameter value is N, the cursor is moved N spaces to the right, if zero or 1, the cursor is moved one space to the right. When attempting to move the cursor to the right of the right margin, the cursor stops at the right margin.

CURSOR POSITION
ESC [Pn;PmH

This sequence moves the active (cursor) position to the position determined by two parameter values, the first specifying the line position and the second specifying the column position. A value of zero or 1 in the first parameter places the cursor in the first line; a value of zero or 1 in the second parameter places the cursor in the first column. Likewise, a value of 15, 10 in the parameters will move the cursor to line 15, column 10. The default condition, with no parameters present, e.g. ESC [H, is equivalent to a Home command, and the cursor will move to the top left corner of the screen, or line 0, column 0.

The numbering of lines depends on the state of the Origin Mode.

CURSOR UP
ESC|PnA

This sequence moves the active (cursor) position upward without altering the column position. The distance moved is determined by the numeric parameter; i.e., if the parameter value is N, the cursor is moved N lines upward, if zero or 1, the cursor is moved one line upward. When attempting to move the cursor above the top margin, the cursor stops at the top margin.

DEVICE ATTRIBUTES
ESC|Pnc

1. A CONTROL sequence with either no parameters or a parameter of 0 is sent by the host to the MIME 100 requesting the MIME 100 to identify itself.
2. The MIME 100 response to this request is generated by a control sequence with the numeric parameters as follows:

OPTION PRESENT

SEQUENCE SENT

MIME 100 No Options	ESC 21;2c
MIME 100 and STP	ESC 21;3c
MIME 100 and GRO	ESC 21;6c
MIME 100, GRO, and STP	ESC 21;7c
GRO - Graphics Option	
STP - Processor Option	

DEVICE STATUS REPORT
ESC|Psr

The general status of the MIME 100 is requested and reported according to the following selective parameter (Ps):

PARAMETER	PARAMETER MEANING
0	Response from MIME 100—Ready, No malfunctions detected (default)
3	Response from MIME 100—Malfunction—retry
5	Command from host—Please report status (using a DSR control sequence)
6	Command from host—Please report active position (using a CR control sequence)

ERASE IN DISPLAY
ESC|Psd

This sequence erases some or all of the characters in the display as determined by the parameter. Any complete line erased by this sequence will return that line to single width mode.

PARAMETER	PARAMETER MEANING
0	Erase from cursor position to the end of the screen (default)
1	Erase from the start of the screen to and including the cursor position
2	Erase all of the display—all the lines are erased and changed to single-width. The cursor does not move.

ERASE IN LINE
ESC|Psk

This sequence erases some or all of the characters in the active line as determined by the parameter.

PARAMETER	PARAMETER MEANING
0	Erase from the cursor position to the end of the line, inclusive (default)
1	Erase from the start of the line to and including the cursor position
2	Erase all of the line, inclusive

HORIZONTAL TABULATION SET
ESC|H

This sequence sets one horizontal stop at the cursor position.

HORIZONTAL AND VERTICAL POSITION
ESC|Pn;Pnf

This sequence moves the active (cursor) position to the position determined by two parameter values, the first specifying the line position and the second specifying the column position. A value of zero or 1 in the first parameter places the cursor in the first line; a value of zero or 1 in the second parameter places the cursor in the first column. Likewise, a value of 15, 10 in the parameters will move the cursor to line 15, column 10. The default condition, with no parameters present, e.g. ESC|f, moves the cursor to the home position. The numbering of lines and columns depends on the reset or set state of the Origin Mode.

INDEX
ESC|D

This sequence causes the cursor to move downward one line without changing the column position. If the cursor is at the bottom margin, a scroll up is performed.

LINE FEED/NEW LINE MODE

Set Mode: ESC[20h
Reset Mode: ESC[20l

The Reset Mode control sequence, e.g., ESC[20l, causes the cursor to move vertically, and the RETURN key (CR) to send the single code CR. The Set Mode control sequence, e.g., ESC[20h, causes the cursor to move to the first position of the following line, and causes the RETURN key to send the two codes (CR,LF). The state of this option upon power up is determined by a "switch" setting. See Section 1.6.3.

NEXT LINE ESC E

This sequence causes the active position to move to the first position on the next line downward. If the active position is at the bottom margin, a scroll up is performed.

SCREEN ALIGNMENT DISPLAY ESC @

The command fills the entire screen area with uppercase E's for screen focus and alignment. This command is used by Micro-Term manufacturing personnel.

ANSI/VT52 MODE

Set Mode: N/A
Reset Mode: ESC[72l

The Reset Mode control sequence, e.g., ESC[72l, causes only VT52 compatible escape sequences to be interpreted and executed. The Set Mode control sequence causes only ANSI-compatible escape and control sequences to be interpreted and executed.

AUTO REPEAT MODE

Set Mode: ESC[78h
Reset Mode: ESC[78l

The Set Mode control sequences cause certain keyboard keys to auto-repeat. The Reset Mode control sequences cause no keyboard keys to auto-repeat.

AUTOWRAP MODE

Set Mode: ESC[77h
Reset Mode: ESC[77l

The Set Mode control sequences, e.g., ESC[77h, cause any displayable characters received when the cursor is at the right margin to advance to the start of the next line, performing a scroll up if required and permitted. The Reset Mode control sequence, e.g., ESC[77l, cause these characters to overwrite any previous characters at the right margin.

CURSOR KEYS MODE

Set Mode: ESC[71h
Reset Mode: ESC[71l

This mode is only effective when the terminal is in keypad application mode and the ANSI/VT52 mode is set. Under these conditions, the Set Mode control sequences will cause the four cursor function keys to send application functions. The Reset Mode control sequences will cause the four cursor function keys to send ANSI cursor control commands. See Section 2.1.5.

COLUMN MODE

Set Mode: ESC[73h
Reset Mode: ESC[73l

The Set Mode control sequences will cause a maximum of 132 columns on the screen. The Reset Mode control sequences will cause a maximum of 80 columns on the screen.

DOUBLE HEIGHT LINE

Top Half: ESC[33
Bottom Half: ESC[34

The line containing the cursor becomes the top or bottom half of a double-height, double-width line, as determined by the control sequences. In order to insure full double-height characters, the control sequences must be used in pairs on adjacent lines, and the same character output must be sent to both lines. If the line was single-width, single-height, all characters to the right of the center of the screen are lost. If the cursor is located to the left of the center screen, it will remain in the same character position; if the cursor is to the right of the center screen, it is moved to the right margin.

DOUBLE-WIDTH LINE

ESC[36

This sequence causes the line that contains the cursor to become double-width, single-height. If the line was single-width, single-height, all characters to the right of the center of the screen are lost. If the cursor is located to the left of the center screen, it will remain in the same character position; if the cursor is to the right of the center screen, it is moved to the right margin.

IDENTIFY TERMINAL

ESC Z

This sequence causes the same response as the ANSI DEVICE ATTRIBUTES. This sequence will not be supported in future terminals, therefore the ANSI device attributes command sequence should be used by any new software.

INTERLACE MODE

Set Mode: ESC[79h
Reset Mode: ESC[79l

The Set Mode control sequence, e.g., ESC[79h, (interlace) causes the video processor to display 480 scan lines per frame if the graphics processor option

has been installed. The Reset Mode control sequence, e.g., ESC[79l (non interlace) causes the video processor to display 240 scan lines per frame. There is no increase in character resolution.

KEYPAD APPLICATION MODE ESC<

The auxiliary keypad keys and cursor control keys will transmit control sequences as defined in Tables 2.1.5 and 2.1.6.

KEYPAD NUMERIC MODE ESC >

The auxiliary keypad keys will send ASCII codes corresponding to the characters on the keys. The cursor control keys will send cursor controls.

LOAD LEDs ESC[Paq

Load the four programmable LED's on the keyboard according to the selective parameter(s).

PARAMETER	PARAMETER MEANING
0	Clear LED's L1 through L4
1	Light L1
2	Light L2
3	Light L3
4	Light L4

The LED numbers are indicated on the keyboard.

ORIGIN MODE Set Mode: ESC[76h Reset Mode: ESC[76l

The Set Mode control sequence, e.g., ESC[76h, causes the origin (home) to be at the upper left character position within the margins. Refer to SET TOP AND BOTTOM MARGINS. Once margins have been defined, the line and column numbers are relative to those margin settings. The cursor is not allowed to be positioned outside the margins.

The Reset Mode control sequence, e.g., ESC[76l, causes the origin (home) to be at the upper-left character position on the screen (column 1, line 1). If margins have been set (refer to SET TOP AND BOTTOM MARGINS), the line and column numbers are independent of these settings. The cursor may be positioned outside the margins with a cursor position or horizontal and vertical position control.

The cursor is moved to the new home position when this mode is set or reset.

Lines and columns are numbered consecutively, with the origin (home) being line 1, column 1 (upper left character position).

RESTORE CURSOR ESC 8

This sequence causes the previously saved cursor position, graphic rendition, and character set to be restored. Refer to SAVE CURSOR.

SAVE CURSOR ESC 7

This sequence causes the cursor position, graphic rendition, and character set to be saved. Refer to RESTORE CURSOR.

SCROLLING MODE Set Mode: ESC[74h Reset Mode: ESC[74l

The Set Mode control sequence, e.g., ESC[75h, causes the scroll to be "smooth" at a maximum rate of six lines per second. The Reset Mode control sequence, e.g., ESC[75l, causes the scroll to "jump" instantaneously.

SCREEN MODE Set Mode: ESC[75h Reset Mode: ESC[75l

The Set Mode control sequence, e.g., ESC[75h, causes the screen to be white with black characters. The Reset Mode, e.g., ESC[75l, causes the screen to be black with white characters.

SET TOP AND BOTTOM MARGINS ESC[Pn;Pnr

This control sequence lets the top and bottom margins to define the scrolling region. The first numeric parameter in the control sequence refers to the first line in the scrolling region; the second numeric parameter refers to the bottom line in the scrolling region. Default is the entire screen, i.e., no margins—the entire screen will scroll. The minimum size of the scrolling region allowed is two lines, i.e., the top margin (line number) must be less than the bottom margin (line number). The cursor is placed in the home position. Refer to ORIGIN MODE.

SINGLE-WIDTH LINE ESC[45

This control sequence causes the line containing the cursor to become single-width, single-height. The cursor remains at the same character position. This is the default condition for all new lines on the screen.

REVERSE INDEX ESC M

This control sequence causes the cursor to move to the same horizontal position on the preceding line. If the cursor position is at the top margin, a scroll down is performed.

RESET TO INITIAL STATE
ESC c

This control sequence resets the MIME 100 to its initial state, i.e. the state it has after it is powered on. This sequence also causes the execution of the power-up self-test.

RESET MODE
ESC(Pa;Ps;...;Pah

This control sequence resets one or more MIME 100 modes as defined by each selective parameter in the parameter string. Each mode to be reset is specified by a separate parameter. Refer to SET MODE.

SET MODE
ESC(Pa;...;Pah

This control sequence sets one or more MIME 100 modes as defined by each selective parameter in the parameter string. Each mode to be set is specified by a separate parameter. A mode is considered set until it is reset by a Reset Mode control sequence. Refer to RESET MODE.

PARAMETER	MODE FUNCTION	"SET"	"RESET"
1	Cursor Key	Application	Cursor
2	ANSI/VT52	ANSI	VT52
3	Column	132	80
4	Scrolling	Smooth	Jump
5	Screen	Reverse	Normal
6	Origin	Relative	Absolute
7	Auto Wrap	Enabled	Disabled
8	Auto Repeat	Enabled	Disabled
20	New Line	New Line	Line Feed

All other parameters are ignored.

SELECT CHARACTER SET

The appropriate G0 and G1 character sets are designated from one of the five possible character sets. The G0 character set is enabled by the code SI (Shift In), while the G1 character set is enabled by the control code S0 (Shift Out).

G0 SETS SEQUENCE	G1 SETS SEQUENCE	MEANING
ESC(A	ESC(A	United Kingdom Set
ESC(B	ESC(B	ASCII Set
ESC(0	ESC(0	Special Graphics
ESC(1	ESC(1	Alternate Character ROM
ESC(2	ESC(2	Standard Character Set
		Alternate Character ROM
		Special Graphics

The United Kingdom and ASCII sets conform to the "ISO international register of character sets to be used with escape sequences". The other sets are private character sets. Special Graphics means that the graphic characters for the codes 6F Hex to 7E Hex are replaced with other characters (see Table 2.3.1). When a specified character set is enabled through the use of the Select Character Set codes SI or S0, that character set will be used until another Select Character Code (SI or S0) is received.

SELECT GRAPHIC RENDITION
ESC(Ps;...;Psm

This control sequence enables the graphic rendition as specified by the parameter(s). All characters transmitted to the MIME 100 following the control sequence are rendered according to the specified parameter(s) until the next occurrence of a Select Graphic Rendition control sequence.

PARAMETER	PARAMETER MEANING
0	Attribute Off (Default)
1	Bold or Increased intensity
4	Underscore
5	Blink
7	Negative (reverse) image
	All other parameter values are ignored.

TABULATION CLEAR
ESC(Psg

This control code sequence causes tabs to be cleared as specified by the numeric parameter. Default value is 0.

PARAMETER	PARAMETER MEANING
0	Clear the horizontal tab stop at the active position.
3	Clear all horizontal tab stops.
	All other parameter values are ignored.

2.9 CONTROL SEQUENCE SUMMARY

2.9.1 ANSI MODE CONTROL SEQUENCES

CURSOR MOVEMENT

Cursor Up	ESC/PnA
Cursor Down	ESC/PnB
Cursor Forward (right)	ESC/PnC
Cursor Backward (left)	ESC/PnD
Direct Cursor Addressing	ESC/Pl;PcH
or	ESC/Pl;PcF
Index	ESC/D
New Line	ESC/E
Reverse Index	ESC/M
Save Cursor and Attributes	ESC/7
Restore Cursor and Attributes	ESC/8

ERASING

From Cursor to End of Line From Beginning of Line	ESC/K or ESC/OK
to Cursor	ESC/LK
Entire Line Containing Cursor	ESC/LK
From Cursor to End of Screen	ESC/U or ESC/OU
From Beginning of Screen	ESC/U
to Cursor	ESC/LU
Entire Screen	ESC/2U

CHARACTER LINE SIZE

Change this line to double-height top half	ESC/3
Change this line to double-height bottom half	ESC/4
Change this line to single-height single-height	ESC/5
Change this line to double-width single-height	ESC/6

CHARACTER ATTRIBUTES

ESC/Ps/Ps/Ps/.../Ps m

Ps refers to a selective parameter. Multiple parameters are separated by the semicolon character (3BH). The parameters are executed in order and have the following meanings:

0 or None	All Attributes Off
1	Bold on
4	Underscore On
5	Blink On
7	Reverse Video On

CHARACTER SETS

CHARACTER SET	G0 DESIGNATOR	G1 DESIGNATOR
United Kingdom (UK)	ESC/A	ESC/A
United States (USASCII)	ESC/B	ESC/B
Special Graphics Characters	ESC/C	ESC/C
and Line Drawing Set	ESC/D	ESC/D
Alternate Character ROM	ESC/E	ESC/E
Alternate Character ROM	ESC/F	ESC/F
Special Graphics Characters	ESC/G	ESC/G

SCROLLING REGION

ESC/Pt;Pb r

Pt is the number of the top line of the scrolling region; Pb is the number of the bottom line of the scrolling region and must be greater than Pt.

TAB STOPS:

Set tab at current column	ESC/H
Clear tab at current column	ESC/G or ESC/Ig
Clear all tabs	ESC/3g

PROGRAMMABLE LEDS

ESC/Ps/Ps;...Ps q

Ps are selective parameters separated by semicolons (3BH) and executed in order, as follows:

0 or None	All LEDs Off
1	L1 On
2	L2 On
3	L3 On
4	L4 On

MODES

MODE NAME	MODE	TO SET SEQUENCE	TO RESET SEQUENCE
Line feed/new line	NEW line	ESC/20h	ESC/20i
Cursor key mode	Cursor	ESC/21h	ESC/21i
ANSI/VT52 mode	ANSI	N/A	ESC/22i
Column mode	132 Col	ESC/23h	ESC/23i
Scrolling mode	Smooth	ESC/24h	ESC/24i
Screen mode	Reverse	ESC/25h	ESC/25i
Origin mode	Relative	ESC/26h	ESC/26i
Wraparound	On	ESC/27h	ESC/27i
Auto repeat	On	ESC/28h	ESC/28i
Keypad mode	Application	ESC =	ESC/29i

REPORTS

Cursor Position Report

Invoked by
Response is

ESC[6n
ESC[Pl;PcR

Pline;Pool

Status Report

Invoked by
Response is

ESC[5n
ESC[On
ESC[3n

(terminal OK)
(terminal not OK)

What Are You

Invoked by
Response is

ESC[c or
ESC[0
ESC[?l;Ps c

Ps is the "option present" parameter with the following meaning:

Ps	Meaning
2	MIME 100, no graphics
3	Processor option (STP)
6	Graphics Processor Option (GPO)
7	STP and GPO

RESET

Reset causes the power-up reset routine to be executed.

ESC c

CONFIDENCE TESTS

Fill Screen with "Ps"
Invoke Tests

ESC[8
ESC[2;9y

Tests exercise ROM, RAM, and EPROM. Tests are repeated indefinitely until failure or power off.

The Hamilton and Area Packet Network

The second meeting of the Hamilton and Area Packet Network will be held at:

Seminar room,
The Canada Centre for Inland Waters,
867 Lakeshore Road,
Burlington, Ontario.

on Sunday, March 8th at 13:00 EST.

Subjects to be thrashed out include Applications, Protocols and ...

Plan to attend and if you have any friends who are really interested in setting into Packet operations, feel free to invite them.

Directions from Toronto:

QEW towards Niagara at Burlington,
exit to the Beach Blvd./Lakeshore Road
Just before the Skyway bridge,
Right at stoplight on Lakeshore Road,
about 1/2 mile on the right, under the
Skyway is C.C.I.W.

From the South:

QEW to Beach Blvd.,
Left at light onto Beach Blvd,
Cross the Lift Bridge at the Canal,
Left at Main entrance into C.C.I.W.,
Just past the bridge.

9-TRIC

OUT

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;RESTORE I/O50K LOC AND ATTR
;RESET DISPLAY SYSTEM

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This bibliography was made by searching the last 6 months of Current Contents, Engineering, Technology and Applied Sciences for the words ALOHA, Network, Packet or Protocol. The number after the sequence number is the issue number of CC the citation was found in. Certain liberties have been taken in leaving out citations found: if the language was Russian, if the network was obviously a TV or Radio broadcast network or if it was a movie titled 'Aloha, Robby and Rose'.

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-A new class of protocols for multiple access in satellite networks,
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-Communications protocol for dual terminals,
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 - INFO II P 357
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 - DEMAND ASSIGNMENT MULTIPLE ACCESS SYSTEM FOR THIN ROUTE NETWORKS
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47. 39
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 - ANALYSIS OF SHARED FINITE STORAGE IN A COMPUTER NETWORK NODE ENVIRONMENT UNDER GENERAL TRAFFIC CONDITIONS
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50. 37
 - A QUEUEING ANALYSIS OF 2/ARO PROTOCOLS
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51. 37
 - PACKET SWITCHING IN RADIO CHANNELS
 - NEW CONFLICT FREE MULTIPLE ACCESS SCHEMES
 - IEEE TRANS ON COMMUN V 28 N 7 P 1015
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 - AN INTERNATIONAL EXPERIMENT IN HIGH-SPEED COMPUTER NETWORKING VIA SATELLITE
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54. 81-6
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DESCRIPTION OF MODEM OF VE3JWV

The heart of the modem are the XR2206 sinewave generator, and the XR2211 phaselock demodulator chips.

The RELL 202 compatible tones used are 1700 and 2200 Herz. Standard RS232 interface levels are used for interfacing with the T.M.C. . The leds for monitoring the interface signals are quite helpful for checking line status.

Another feature is the analog switch MC14016 which turns off the modem output when you want to talk over the radio (PTT pressed)

The microphone is here continuously connected and does not interfere with packet transmissions. Also if the T.M.C. wants to transmit a packet this packet will wait until you let the mike button go (CTS being delayed).

The dip switch allows setting the modem for half duplex (normal mode) or full duplex and allows easy testing. Also S7

allows setting of an extra long CTS delay for some of our fellow slowokes with relay switching.

CONSTRUCTION AND ADJUSTMENT :

The modem is build on a small plussable card with a 22 pin edge connector from my local radio shack and attached to the same power supply as the T.M.C. .

The modem connects to the radio with a shielded cable that, in my case plugs into the existing socket of my ICOM 22S which is slightly modified to accommodate the modem.

The adjustments are very simple. First set up the XR2206 tones using a frequency counter, adjusting f1 for 1200 and f2 for 2200 Herz. Then loop the signal back into the demodulator (S9=0, S2=1, S1=0). Now adjust the loop frequency so that with the T.M.C. sending floss (normal initialized condition) till the RXD looks identical to TXD.

The asymmetry is not critical and could be substituted with a permanent resistor 150 ohm.

The 50 K pot. should be adjusted for proper deviation here approximately 3 Kc. This can easily be checked with a scope at the friend's receiver.

The setting of the request to send delay (74121) should be done experimentally. For a ris with solid state switching a value around 100 msec is probably ok, for relay switching close S7 and try around 400 msec.

Good luck, see you on PACKET RADIO.

(JOHANN C. VANDEN BERG)

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- 60. 81-6
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-WISSENSCHAFTLICHE BERICHTE AEG-TELEFUNKEN
V 53 N 3 P 111, 126

(EDITOR'S NOTE: THE DUMP
WILL APPEAR IN THE
NEXT BUILDING)


```

*****
TIP THE TRAP MODULE
*****
THIS PROGRAM IS ENTERED ON ACTIVATING
THE TRAP SWITCH OF THE T.M.C.

THE PROGRAM SAVES THE REGISTERS AND
INITIALIZES THE W0R250 HOST INTERFACE

THEN THE T.M.C. WILL SIT WITH THE
INDICATOR LED FLASHING WAITING FOR
THE HOST TO SEND THE TRANSFER REQUEST

WHEN THE TRANSFER REQUEST IS RECEIVED
ALL BK BYTES OF MEMORY ARE SENT TO THE
HOST AND THE TRANSFER REQUEST LOOP IS
ENTERED AGAIN.

TO TERMINATE HIT THE T.M.C. RESET

01/01/81 J.C.WARDEN BERG
01/10/81 ADDED FLASHING LED
*****

.PARS
;
;
DEFINE RIM = C.BYTE 20H)
DEFINE SIM = C.BYTE 30H)

;
;
UWAT = ON 18250 BASE ADDRESS
UWATH = OHM 18 DATA BITS, 1ST O/P BIT, ODD PARITY
DIVISOR = B FOR 19,200 BAUD

0000 RE50 = UWAT
0001 RE61 = UWAT+1
0002 RE62 = UWAT+2
0003 RE63 = UWAT+3
0004 RE64 = UWAT+4
0005 RE65 = UWAT+5
0006 RE66 = UWAT+6

;
;
LOC 0C00H 1$TANT OF EPROM
;
; FILL FIRST PART OF EPROM WITH 0FFH
;
OC00H FFFF FFFF
LOC000H:
OC00H OFFFH,OFFFH,OFFFH,OFFFH,OFFFH

```

[illegible][illegible]


```

ROUTINE TO DRIVE JART INTERFACE
GENERAL PACKET CONTROL  VERS 3.1
ORG 07500H  JASSIGN START
SPORT EQU 0F0H  JDATA PORT OF JART
SPORT EQU 0FEH  JSTATUS & CONTROL PORT
BASIC EQU 0  JCOL START BASIC TO FIX PTRS
TSORN EQU 3000H  JTOP OF SCREEN
BLANK EQU 20H  JASCII BLANK CODE
CURSOR EQU 4020H  JCURSOR POINTER
ENTER EQU 0DH  JENTER KEY
BREAK EQU 1  JBREAK KEY
LF EQU 0AH  JDOWN ARROW KEY
VIDEO EQU 34H  JBASIC SCREEN HANDLER
LINE2 EQU 3040H  J2ND LINE OF SCREEN
KEYS EQU 2BH  JBASIC KEYBOARD SCAN
JSCORE EQU 5FH  JUNDERSCORE CURSOR CHAR.
CLEAR EQU 1FH  JCODE TO CLEAR SCREEN
FFFEED EQU 0CH  JFORM FEED CODE
ESC EQU 1BH  JESCAPE CODE

```

GENERAL PACKET CONTROL PROGRAM

```

PACK CALL INIT  JSET JP JART
CALL KLEAR  JCLEAR THE SCREEN
LD HL,MESS
LD DE,TSORN
LD BC,MESS1-MESS
LDIR
LD HL,LINE2  JSET CURSOR TO LINE 2
LD A,JSCORE  JPJT CURSOR ON SCREEN
LD (HL),A
LD (CURSOR),HL

```

SETUP IS FINISHED. GO INTO GENERAL OPERATION

```

CHKCHK CALL RECV  JHAS ANYTHING BEEN RECEIVED
JR NZ,SHROV  JIF SO, SHOW IT
CALL KEYS  JANYTHING TO SEND?
OR A
JR Z,CHKCHK
CP BREAK  JDOES USER WANT OUT
JF Z,BASIC
CP CLEAR  JIS IT A CLEAR SCREEN REQUEST
JR NZ,TECO
CALL KLEAR  JCLEAR THE SCREEN
JR CHKCHK
TECO CP ESC  JTEST FOR ESCAPE CODE
JR NZ,SHCHAR
LD A,FFFEED  JSUB FORM FEED
SHCHAR CALL KSEND  JSEND IT
CP ENTER  JIS IT THE ENTER KEY
JR NZ,CHKCHK
LD A,LF  JISSUE SEND REQUEST
SEND CALL KSEND
JR CHKCHK
SHROV PUSH AF  JSAVE FLAGS
CALL C,ERROR
POP AF
PUSH AF
CALL M,FERROR
POP AF
CP LF  JCHECK FOR A FORM FEED
JR Z,CHKCHK  JIF SO, IGNORE IT.
CP FFEED  JIS IT A FORM FEED?
JR NZ,SHOW
CALL KLEAR  JCLEAR THE SCREEN
JR CHKCHK
SHOW CALL VIDEO
LD HL,(CURSOR)
LD A,40H  JCHECK FOR SCREEN OVERFLOW
CP H
JR NZ,CHKCHK
LD HL,TSORN
LD (CURSOR),HL
JR CHKCHK

```

```

ERROR LD HL,MESS1
LD DE,TSORN
LD BC,MESS2-MESS1
LDIR
RET
FERROR LD HL,MESS2
LD DE,TSORN+20H
LD BC,COUNT-MESS2
LDIR
RET
INIT LD A,6BH  JB DATA & 1 STOP BIT, NO PARIT
OUT (SPORT),A
IN A,(SPORT)  JCLEAR STATUS REG.
VFRAME LD A,24H  JINITIALIZE BYTE COUNTER
LD (COUNT),A
RET
KSEND CALL SEND  JSEND BYTE
PUSH AF  JSAVE IT
CP LF  JHAS IT A SEND REQUEST?
JR NZ,KCHAR
LD A,LF  JAUTOMATIC SEND REQUEST
SEND CALL SEND
CALL VFRAME  JRESET BYTE COUNT
JR RSTOR
LD A,(COUNT)  JGET BYTE COUNTER
DEC A  JCOUNT IT
LD (COUNT),A  JPJT IT BACK
JR Z,SEND  JIF COUNT IS ZERO SEND BUFFER
RSTOR POP AF  JGET BYTE BACK
RET
SEND PUSH AF  JSAVE BYTE
IN A,(SPORT)  JCHECK IF XMIT BUF IS FULL
BIT 7,A  JCHECK 'TBE'
JR Z,WAIT  JWAIT FOR IT TO EMPTY
POP AF  JGET BYTE BACK
OUT (SPORT),A  JSEND IT
RET
RECV IN A,(SPORT)  JCHECK IF DATA AVAILABLE
BIT 0,A  JCHECK BIT
JR Z  JRETURN IF NO DATA AVAIL.
LD B,A  JSAVE CONTENTS
XOR A  JZERO REG A
LD 1,B  JCHECK FOR OVER RJN
JR Z,CHKFRM  JIF OVER RJN SET CARRY
LD 0,A  JFOR CARRY FLAG
CHKFRM BIT 2,B  JCHECK FRAMING
JR Z,SETDAT  JIF FRAMING ERROR
SET 7,A  JFOR SIGN FLAG
SETDAT LD C,A
IN A,(SPORT)  JGET DATA BYTE
LD B,A  JMOVE IT TO SAVE POS.
BC  JMOVE FLAG STATUS
POP AF  JTO FLAG REG
RET
KLEAR LD HL,TSORN  JROUTINE TO CLEAR SCREEN
LD (CURSOR),HL  JSET POINTER
LD A,BLANK
LD (HL),A
LD DE,TSORN+1
LD BC,400H
LDIR
RET
MESS DEFM *PACK MODE READY
MESS1 DEFM *** OVER RJN ERROR ***
MESS2 DEFM *** FRAMING ERROR ***
COUNT DEFM 0
END PACK

```


ROUTINE TO SEND FROM BASIC TO THE TTY PACKET
 TIP. WARNING. CHARACTERS WILL BE DROPPED UNLESS
 YOU MAKE SURE TO NEVER FULLY FILL THE SEND BUFFER.
 EXAMPLE: ONLY "LIST" A FEW LINES AT A TIME
 "LPRINT" WITH LOTS OF DEAD TIME.

BY 3LENN VE3OSP VER 2.1

OR3	7F30H	TTY ENTRY ADDRESS
SPORT	0F0H	UART DATA PORT
PADDR	0FEH	UART CONTROL & STATUS
LF	4026H	PRINTER ADDRESS FOR BASIC
BASIC	0AH	LINE FEED CODE
	1A19H	WARM START

SET JART FOR 1200 B4JD

LPACK	LD	A,63H	18 BIT DATA 1 STOP VP
	OUT	(SPORT),A	
	IN	A,(SPORT)	10CLEAR RX BUFFER
	LD	HL,SEND	10SET BASIC POINTER
	LD	(PADDR),HL	
	JP	BASIC	10GIVE CONTROL TO BASIC

NOTE: DATA FROM THE TIP IS IGNORED

SEND	LD	A,C	10SET BYTE
	OP	00H	10CR?
	JR	Z,CARET	
	OP	20H	10RET IF A CONTROL CODE
	RET	M	
	CALL	BYOUT	10SEND THE BYTE
	PUSH	AF	10SAVE REGS
	PUSH	BC	

WAIT FOR EITHER A CHAR TO ECHO -OR- A TIME OUT

OOP	LD	BC,0000H	10SET MAX DELAY COUNT
	IN	A,(SPORT)	10SEE IF CHAR ECHOED
	BIT	7,A	
	JR	NZ,COUNT	
	DEC	C	
	JR	NZ,LOOP	
ONT	POP	BC	
	POP	AF	
	RET		

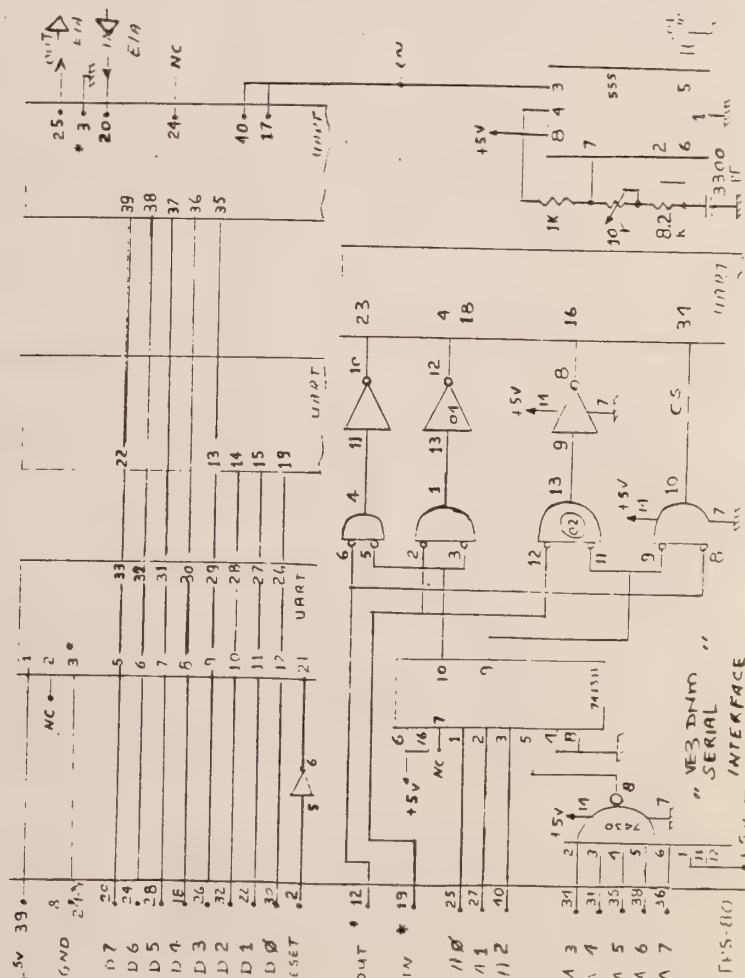
SUBROUTINES OF THE MAIN SUBROUTINE

LINE	LD	A,00H	
RET	CALL	BYOUT	10PRINT CR
	LD	A,LF	10ADD A LF
TOJT	PUSH	AF	10SAVE BYTE
BIT	IN	A,(SPORT)	10SET TX STATUS
	BIT	7,A	10IS TX BUFFER EMPTY?
	JR	Z,WAIT	
	POP	AF	
	OUT	(SPORT),A	10PRINT THE BYTE
	RET		
END	LPACK		

ROUTINE TO SEND AT 110 B4JD FROM THE MD AT 110 B4JD			
OR3	7F30H	TTY ENTRY ADDRESS	
SPORT	0F0H	UART DATA PORT	
PADDR	0FEH	UART CONTROL & STATUS	
BASIC	4026H	ADDRESS PTR FOR BASIC	
	1A19H	WARM START	

TTY	LD	A,74H	10SET 8 BIT DATA 2 STOP VP
	OUT	(SPORT),A	
	IN	A,(SPORT)	10CLEAR RX BUFFER
	LD	HL,SEND	10SET BASIC POINTER
	LD	(PADDR),HL	
	JP	BASIC	10GIVE CONTROL TO BASIC
SEND	LD	A,C	10SET BYTE
	OP	00H	10CR?
	JR	Z,CARET	
	OP	20H	10RET IF A CONTROL CODE
	RET	M	
	PUSH	AF	
	LD	A,(COLNO)	10FIND OUT WHAT COL. #
	OR	A	
	CALL	Z,VLINE	10IF FULL THEN NEW LINE
	POP	AF	
	CALL	BYOUT	10PRINT THE BYTE
	PUSH	HL	
	LD	HL,COLNO	10COUNT THE BYTE
	DEC	(HL)	
	POP	HL	
	LD	A,C	
	RET		

VLINE	LD	A,00H	10ISSUE OR
CARET	CALL	BYOUT	10PRINT CR
	LD	A,74H	10SET NEW LINE COUNT
	LD	(COLNO),A	
	LD	A,0AH	10ADD A LF
	PUSH	AF	10SAVE BYTE
BYOUT	IN	A,(SPORT)	10SET TX STATUS
WAIT	BIT	7,A	10IS TX BUFFER EMPTY?
	JR	Z,WAIT	
	POP	AF	10IF EMPTY SO
	OUT	(SPORT),A	10PRINT THE BYTE
	RET		
COLNO	DEFB	20	
END	TTY		



MAR 4-6 1980, ZURICH

PACKET SPEECH FOR LAND MOBILE CHANNELS

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ABSTRACT

This paper describes the results of a study to evaluate the feasibility of applying packet speech concepts to land mobile systems.

One particular random access technique (MPCSA) is investigated and it is shown that for typical values of the system parameters packet speech concepts favourably to conventional trunked land mobile systems.

An expression is derived for the maximum number of active users that can be supported by a single 30 KHz channel, as a function of the voice digitization rate and the lost packet level. It is further shown that the number of base-to-mobile channels does not need to be equal to the number of mobile-to-base channels.

INTRODUCTION

In the past few years considerable interest has been demonstrated in digital voice techniques /1, 2/, packet radio random access schemes /3, 4/ and cellular structures for land mobile communications /5, 6/.

A number of experiments have been carried out over the ALPACAT /1/, to demonstrate the feasibility of transmitting packetized voice. Also the integration of packetized data and packetized voice has been advocated by some researchers /8/ who have attempted to quantify the performance measures of such integrated networks and provide some design guidelines.

Closely related to the idea of packetized voice is the concept of speech interpolation which permits a number of voice sources to share a number of channels through voice-activated switching by taking advantage of the gaps or pauses that naturally occur during a conversation. One of the earliest /9/ speech interpolation systems known as TASI (Time Assignment Speech Interpolation) was a pure analog system whose performance parameter was the fraction of speech lost due to freeze-outs. A freeze-out occurs when a given talker finds no channel idle. Recently a digital version of speech interpolation known as DSI has been advocated in particular for satellite circuits /10/.

On the other hand a considerable body of knowledge /11, 12, 13/ exists for the so-called packet radio systems whose potential have been exploited uniquely for data transmission purposes.

Packet radio concepts have also been applied to the control channels of the Chicago /14/ and Tokyo /15/ cellular systems. It has also been suggested by Moray /16/ that packet radio concepts could be efficiently applied to the signalling channels of the wireless mobile services.

However, no attempt has so far been made to apply the concepts of digital speech interpolation and packetized voice to increase the traffic carrying capacity of land mobile systems.

In this paper we show that for typical values of the system parameters, a land mobile system that employs packetized voice concepts, compares favourably to the conventional trunked land mobile system. We derive an expression for the maximum number of active users that can be supported by a single 30 KHz uplink channel (mobile to base) and show that the number of downlink (base to mobile) channels does not need to be equal to the number of uplink channels.

OPERATIONAL CONCEPT

Consider a conventional land mobile system where a large number of speech sources want to exchange voice communication through a finite number of radio channels available at a given base station. If we assume an Erlang B traffic model, the grade of service or blocking probability is given by:

$$P_B = \frac{M!}{M! - L!} \frac{P^L}{L!} \quad (1)$$

where "M" is the traffic offered to the group of "M" channels by the population of users. The total traffic carried by these channels is therefore:

$$P_C = P (1 - P_B) \quad (2)$$

The number of users that can be supported is easily obtained by dividing "P" by the traffic load per voice source. If as an example we consider 100 channels, a blocking probability of 0.02 and a load per voice source of 0.02 erlangs, we conclude that approximately 49 users per channel could be supported.

The same channels could be shared, perhaps more efficiently, by (1) taking advantage of the

statistical characteristics of voice (2) digitizing the talker and encoding them into packets of fixed size and (3) transmitting these packets at high speed. As shown in the model of Figure (1) a very large number of speech sources could share the channel if it were possible to perfectly utilize the pauses in the conversations. It is also clear from the same figure that in such an ideal case the number of speech sources that could share the channel would be a function of the talker length distribution, the pause length distribution, the encoding rate (R), the packet size (B), the overhead (b) and the transmission speed (C).

In practice it is however, virtually impossible to perfectly schedule the activity of the channel without resorting to a centralized control mechanism. A possible alternative is to use a form of distributed control, which simply means that a packet from a given user could be prevented from being transmitted if the channel happened to be busy. If however at some point in time the channel is idle, it is quite possible that nobody else is in the process of transmitting a voice packet.

Fortunately enough such access protocols have already been extensively studied /17/. In the context of packetized data transmission, one of these protocols known as the Non-Persistent Carrier Sense Multiple Access (NPMCSMA) scheme operates as follows:

- If a terminal has a packet ready for transmission it senses the channel and if the channel is sensed idle, the packet is transmitted. It can however, collide with some other packet during a time window, which is related to the propagation delay between terminals.

- If the channel is sensed busy the terminal does not persist in sending the packet and simply retransmits the transmission of the packet according to some random delay distribution. At this new point in time, the channel is sensed again and the same procedure is repeated.

- If a terminal learns that a packet collided with some other packet, it reattempts a retransmission according to the above procedure.

Defining by "N" the average number of packets generated per packet transmission time and by "G" the average number of new and retransmitted packets per packet transmission time it can be shown /17/ that "N" and "G" are related by:

$$S = \frac{G \cdot e^{-G}}{G(1 - 2G) + e^{-G}} \quad (3)$$

where "G" is the normalized propagation delay is given by:

$$G = \frac{T}{P} \quad (4)$$

where "T" is the one-way propagation delay between any two terminals and "P" is the packet transmission time. It can also be shown /17/ that when a packet is ready for transmission, the probability "G" that the channel is busy is given by:

$$G = \frac{G(1 - G) - 1 + e^{-G}}{G(1 - 2G) + e^{-G}} \quad (5)$$

One of the characteristics of the NPMCSMA random access scheme is that not all arrivals result in actual transmission. Some packets will be blocked because of channel unavailability and only those that find the channel idle will actually be transmitted. Following the notation introduced in /17/ we will denote by "n" the actual rate of transmitted packets. Hence we can write:

$$n = G(1 - G) \quad (6)$$

It is customary to define the probability of successful transmission by the ratio "n/G". However, since "G" includes packets that were not transmitted but were merely blocked prior to transmission, we believe it is more appropriate to define the probability of successful transmission by the ratio "n/N". From equations (3), (5) and (6) we obtain:

$$n = \frac{G}{1 + G} \quad (7)$$

For the case of voice packets we suggest a modification of the NPMCSMA protocol, namely we do not attempt to retransmit a packet that has collided. This implies that, in the above equations, we should interpret "G" as the rate of new and retransmitted packets, "N" as the rate of transmitted packets and "n" as the rate of successfully transmitted packets.

SPEECH MODEL

In the previous section we have given expressions for the probability of sensing a busy channel and for the probability of successful packet transmission, in terms of "N" the normalized (offered) channel traffic. We must now relate "N" to the calling patterns of the voice sources as well as to their talker statistical properties.

Consider a sequence of talker and pause characteristic of the speech pattern of a normal user and assume that the talker as well as the pause are exponentially distributed /18/ with means "1/μ" and "1/ν". Hence the probability density functions for the random variables "T" and "P" are given by:

$$f_T(t) = \frac{1}{T} \exp\left(-\frac{t}{T}\right) \quad (8)$$

and

$$f_P(p) = \frac{1}{P} \exp\left(-\frac{p}{P}\right) \quad (9)$$

We assume that the digitally encoded talker and pause are broken up into "n" packets of length "B". If the voice digitization rate is denoted by "R", we have:

$$P (RT \leq nB) = 1 - \exp\left(-\frac{nB}{RT}\right) \quad (10)$$

Hence, the probability that exactly "n" packets will be needed is:

$$P \left[(n-1)B < RT \leq nB \right] = \exp\left(-\frac{(n-1)B}{RT}\right) \cdot \exp\left(-\frac{nB}{RT}\right) \quad (11)$$

the sending time "t_g" we will discard a packet if after "k" sensing points an idle channel was not found.

Since, "q", the probability that a given channel is busy, is the same for all channels, the probability "q" of being forced to discard the packet is given by:

$$q = q^k \quad k=1,2,\dots \quad (17)$$

Naturally the probability (1-q) of being able to transmit the packet within the allowable time period is given by:

$$(1-q) = \sum_{k=1}^{\infty} q^{k-1} (1-q) = 1-q^k \quad (18)$$

An important question can now be raised concerning the number of discarded and collided packets. Since we do not propose that collided packets be retransmitted, we would like to determine the fraction of lost packets (discarded and collided). A given packet within a talkspurt can be discarded with probability "q", and a transmitted packet can collide with probability "1-q", hence the fraction of lost packets in the uplink channel is given by:

$$\theta_u = q + (1-q)(1-q) \quad (19)$$

To determine the fraction of lost packets on the downlink channel we assume that packets arrive at the base station via "h" channels. Denoting by "s" the probability of a successful arrival over any one of these channels, if there are "h" downlink channels (L-H) the fraction of lost packets, "θ_d", is obtained from:

$$\theta_d = \frac{\binom{h}{1} s^1}{\sum_{i=1}^h \binom{h}{i} s^i} \quad (20)$$

To obtain the above expression we have assumed that no buffering is provided at the base station. This is a reasonable assumption since it is well-known that large delays in packet voice transmission will be intolerable. It should also be emphasized that so far we have dealt with what we could call "incoherent" traffic. Indeed we have assumed that communications take place between mobile terminals, and have excluded communications coming in from or addressed to land terminals.

END-TO-END DELAY

For those packets that were successfully transmitted from origin to destination through the base station we can easily derive an upper bound for the maximum end-to-end delay as follows:

$$D = \frac{3B}{R} \quad (21)$$

where the first B/R is the packetization delay, the second B/R is the sum of the maximum pre-transmission delay and the transmission time, and the third B/R is the depacketization delay. Note that we have ignored in the above expression any processing delays that take place at the base station.

Thus the mean number of packets per digitally encoded talkspurt is obtained as:

$$R = (1 - \exp(-\frac{B}{RT}))^{-1} \approx \frac{RT}{B} \quad (12)$$

It is well-known that the speech source activity ratio, "a", can be defined as the ratio of the average talkspurt duration to the sum of the average talkspurt duration and average pause duration, as follows:

$$a = \frac{T}{T + P} \quad (13)$$

The source activity ratio can also be interpreted as the probability that an active speech source is issuing a talkspurt at some random time. Studies [16] have shown that "a" is typically of the order of 0.4.

Since each packet generated during a talkspurt must be identified by a header of size "h" the time required to transmit a packet is given by:

$$T_p = \frac{B+h}{C} \quad (14)$$

where "C" denotes the channel transmission rate.

Finally denoting by "η" the offered load (in Erlangs) per voice source during the busy hour, we obtain the following expression for the normalized offered traffic per source:

$$h = \eta a \frac{B}{C} + \frac{B+h}{C} \quad (15)$$

Since the total traffic offered to each of the "h" channels by a population of "M" voice sources is given by:

$$M = \eta a \frac{B}{C} + \frac{B+h}{C} \cdot M \quad (16)$$

TRAFFIC MODEL

Consider Figure (1) where we see that during a talkspurt, packets "arrive" in a very regular manner. Indeed if each packet contains "B" bits, there will be a packet arrival every "B/R" units of time. This inter packet arrival time corresponds to the so-called packetization delay, that is the time required to form a packet of "B" bits.

Upon arrival of the first packet of a talkspurt to the buffer of the radio terminal, the process of selecting a channel for transmission, is initiated. Suppose that one of the "h" available channels is selected. After a period of time "t_g", the logic unit within the transmitter will decide whether or not the channel is busy. If that particular channel is found to be idle, a header is attached to the packet currently in the buffer and a packet of size "B+h" is transmitted. If the channel is sensed busy, another channel is selected at random among the available "h" channels and the process is repeated. Since the terminal has a buffer capable of containing a single packet and since the interarrival time of two consecutive packets is equal to B/R, a packet currently in the buffer that does not find an idle channel within this period of time is discarded.

More specifically if we divide the period of time (B/R - T_p) into "k" slots each equal to

DISCUSSION AND RESULTS

Before applying the previous equations to a specific set of system parameters it is worthwhile to mention some of the factors that can play a role in selecting such parameters.

Measurements carried out on the characteristics of speech during conversations have shown that human speech is bursty in nature. Brady [16] among others has confirmed that the actual channel utilization during a one-way conversation is only about 40%. He has further shown that the exponential distribution fits reasonably well the distribution of talkspurt lengths with a mean value of about 1.3 sec. On the other hand, results [2, 19] obtained to date on the transmission of packetized speech in the ARPANET indicate that to maintain a high quality speech it is necessary to ensure that:

- a nearly synchronous voice output is generated by the receiver
- end-to-end network delays do not exceed 250 msec.

Moreover, packets of lengths varying from 10 to 50 msec of speech intelligibility can be lost without seriously affecting the voice quality output and degradation begins to be observed when the fraction of lost packets exceeds a certain level, which is a function of the redundancy of the speech signal. According to some of the published data, the tolerable fraction of lost packets varies from 0.35 at low digitization rates to probably 50% at high digitization rates. Hence denoting by "θ" the total fraction of lost packets we have from equations (19) and (20):

$$\theta = \theta_u + \theta_d \leq \begin{cases} 0.005 \text{ for } R \approx 2.4 \text{ kbps} \\ 0.2 \text{ for } R \approx 16 \text{ kbps} \\ 0.5 \text{ for } R \approx 32 \text{ kbps} \end{cases} \quad (22)$$

From equation (16) it can be clearly seen that "η" the normalized offered traffic is highly dependent on the channel transmission rate "C". It is also clear that both the probability "q" of discarding a packet and the ability "1-q" of losing a packet, obtainable from Figures (3) and (4), are highly dependent on the values of "q" and "η". Hence by increasing the channel transmission rate we are clearly increasing the system efficiency measured in terms of the fraction of lost packets. There is however a practical upper bound on the transmission rate that can be derived from a 30 KHz land mobile channel. Indeed a number of modulation schemes that have been devised recently [20], suggest that transmission speeds of the order of 40 kbps, can be achieved over a 30 KHz channel.

Another parameter that can influence the system performance is the packet overhead. For the purposes of our analysis we will assume that an abbreviated header of 16 bits [21] is all that is required to properly address the packets.

Finally the last parameter of crucial importance is the time "t_g" required to sense a channel. As we have mentioned above, (B/R - T_p) the period of time during which channels can be sensed is divided into a number of sensing slots of length "t_g". Hence the maximum number of sensing slots is given by:

$$K = \frac{B/R - T_p}{t_g} \quad (23)$$

which, for all practical purposes is a very large number. Indeed from Table 1, if we assume a channel speed of 40 kbps and a sensing time of 0.03 msec, we see that in the worst case "K" varies from about 53 (R=32 kbps and T_p=0.4 msec) to about 320 (R=32 kbps and T_p=0.4 msec). This immediately implies that for values of "q" below 0.5, the probability "a" of discarding a packet can be ignored.

The values contained in Table 1 which were obtained for two temporal packet lengths of 10 msec and 50 msec (with m=0.02 and m=0.4) indicate, as expected, that for a given channel speed (40 kbps), as the voice digitization rate is increased, the average normalized offered traffic per voice source is also increased. This suggests that in order to increase the maximum number of voice sources that can be supported by the system, the voice digitization rate should be kept as low as possible. However, as indicated above the tolerable fraction of lost packets is a function of the speech redundancy which is quite low for low digitization rates. Since the tolerable lost packet level decreases faster than the normalized offered traffic per voice source, there is little advantage in decreasing the voice digitization rate beyond a certain value. It will be apparent that rather on the contrary, the voice digitization rate should be kept above 16 kbps. Consider a single radio channel supporting an amount of traffic "η" given by equation (16). If we assume that "θ_d", the fraction of packets lost on the downlink channel is negligible, the total fraction of lost packets when a + 0, will be:

$$\theta \approx \theta_u = a + (1-a)(1-q) \approx 1-q \quad (24)$$

Now from Figure (4) we see that, for δ = 0.1, "η" is related to "q" by:

$$\eta \leq \begin{cases} 1 & \text{for } \xi \geq 0.5 \\ 0.55 & \text{for } \xi \geq 0.8 \\ 0.03 & \text{for } \xi \geq 0.995 \end{cases} \quad (25)$$

which implies from (22) and the data of Table 1, that the maximum number of voice sources that can be supported is given by:

$$M \leq \begin{cases} \frac{1}{h} = 148 & \text{for } R \approx 32 \text{ kbps} \\ \frac{0.55}{h} = 156 & \text{for } R \approx 16 \text{ kbps} \\ \frac{0.03}{h} = 37 & \text{for } R \approx 2.4 \text{ kbps} \end{cases} \quad (26)$$

Note that to derive the above numbers we have assumed a packet duration of 10 msec. For packet durations of 50 msec there is a slight increase in the number of users. However, from a delay point of view it is preferable as indicated by equation (21) to keep the packets as short as possible. Note also that in practice δ can be smaller than 0.1. If for instance we take a one-way propagation delay of 54 msec and a packet transmission time of 1 msec we obtain δ=0.054. This implies in particular, that for R=2.4 kbps the maximum number of voice sources that can be supported exceeds the number given by equation (26).

In the case where we have "M" uplink channels, if we assume that the traffic is evenly distributed among these channels, the total number of voice sources that can be supported, is obtained by multiplying the results of equation (26)

by η . Since we want to minimize the fraction of lost packets on the downlink channels, it is essential that for a given value of η , we select the appropriate number N_d of downlink channels. As an example assume that the distribution rate η is equal to 16 hops and that we can tolerate a total fraction of lost packets of the order of 20%. From Figure (4) (with $\delta=0.1$) we find that η should be less than 0.35. Hence the average successful traffic will be 50.44. Using equation (20) with $N_d=50$, we see from Figure (5) that in order to keep η_d below 0.001, the number N_u of required downlink channels is of the order of 27. We can then achieve a spectrum saving of the order of 46% which for 30 KHz channels represents about 0.69 MHz. Additional savings in spectrum can be obtained by allowing η_d to increase while keeping η below 50%.

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CONCLUSION

Based on the results discussed above it appears that the concept of packet speech can be advantageously applied to land mobile channels. We have shown that for a given number of uplink channels, the maximum number of voice sources that can be supported will, under some assumptions, attain a value of ≈ 150 M which is a threshold increase over what can be achieved with conventional analog land mobile channels. We have further shown that, as opposed to the conventional land mobile system where the same amount of spectrum is allocated in both directions, in a packet speech system the amount of spectrum required for the downlink channels represents about 50% of what is required in a comparable analog land mobile system.

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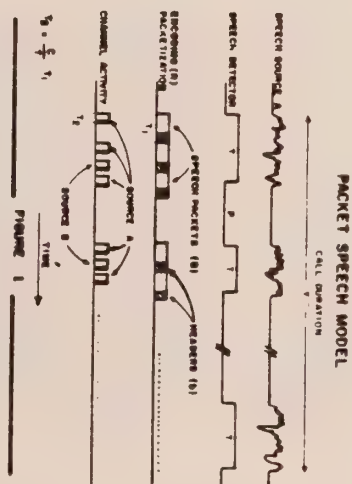


FIGURE 1

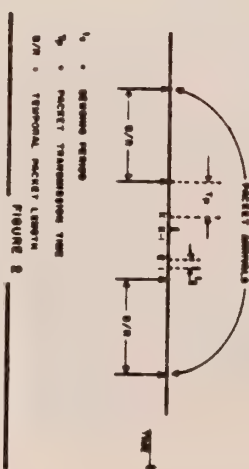


FIGURE 2

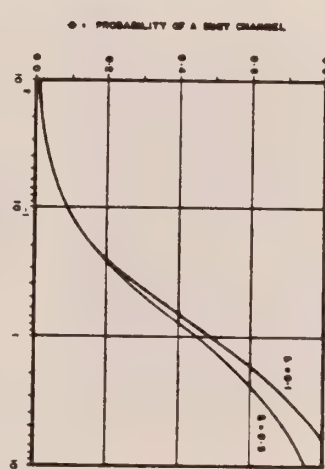


FIGURE 3

G (Erl)	G/G = 10 dB				G/G = 00 dB			
	0.010	0.010	0.010	0.010	0.010	0.010	0.010	0.010
0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
0.1	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
0.2	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
0.3	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
0.4	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
0.5	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
0.6	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
0.7	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
0.8	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
0.9	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
1.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
1.1	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
1.2	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
1.3	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
1.4	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
1.5	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
1.6	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
1.7	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
1.8	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
1.9	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
2.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0

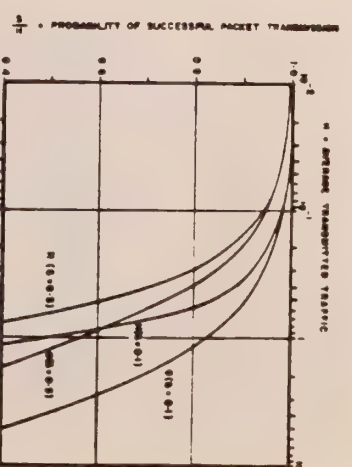


FIGURE 4

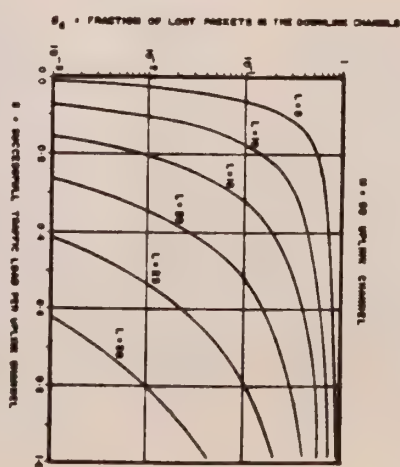


FIGURE 5

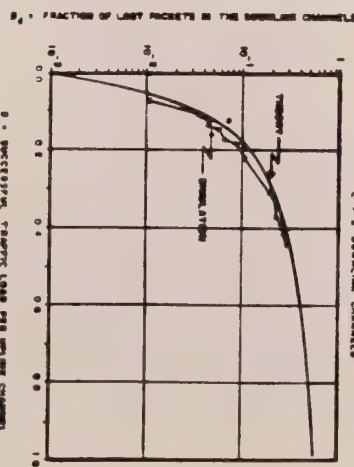


FIGURE 6

ALOHA packet broadcasting—A retrospect

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INTRODUCTION

Packet broadcasting is a technique whereby data is sent from one node in a net to another by attaching address information to the data to form a packet—typically from 30 to 1000 bits in length. The packet is then broadcast over a communication channel which is shared by a large number of nodes in the net; as the packet is received by these nodes the address is scanned and the packet is accepted by the proper addressee (or addressees) and ignored by the others. The physical communication channel employed by a packet broadcasting net can be a ground based radio channel, a satellite transponder or a cable.

Packet broadcasting networks can achieve the same efficiencies as packet switched networks,¹ but in addition they have special advantages for local distribution data networks² and for data networks using satellite channels.³ In this paper we concentrate on those characteristics which are of interest for a local distribution data network. In particular, we discuss the lessons learned in the design and implementation of the ALOHANET, a packet broadcasting radio network in operation at the University of Hawaii since 1970. A number of design issues which arose in the construction of the system are defined, our solutions are explained, and in some cases they are justified. The lessons learned from the ALOHANET are used to indicate how such a radio packet broadcasting system might best be built using the technology available in 1975.

In the next section a brief description of the ALOHANET and its rationale is given. This is followed by a detailed discussion of the major system protocol choices that have evolved, pointing out some related theoretical work where appropriate. Choices concerning the design of the radio communication subsystem are then examined, followed by an evolutionary view of the important impact microcomputer technology has had on the user interface design and resulting system capabilities. The concluding section summarizes our present views with respect to the basic system configuration and properties of packet broadcasting nets.

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THE ALOHANET

The ALOHANET is the first system which successfully utilized the packet broadcasting concept for on-line access of a central computer via radio. Its primary purpose is to provide inexpensive access to one or more time-sharing systems by a large number of terminal users, typically in the hundreds. However, it also allows user-to-user communication within the net and is evolving toward use in a more generally-oriented computer communications environment.

Operation

The present network configuration makes use of a broadcast channel for only one direction of traffic flow. (As we shall see in later sections, the lack of a broadcast capability in the other direction has seriously handicapped the development of effective protocols in certain areas.) Two 100 KHz channels are used in the UHF band—a random access channel for user-to-computer communication at 407.350 MHz and a broadcast channel at 413.475 MHz for computer-to-user messages. The original system was configured as a star network, allowing only a central node to receive transmissions in the random access channel; all users received each transmission made by the central node in the broadcast channel. Recently the addition of ALOHA repeaters has generalized the network structure.

A block diagram of the present operational ALOHANET is shown in Figure 1. The central communications processor of the net is an HP 2100 minicomputer (32K of core, 16 bit words) called the MENEHUNE (Hawaiian for IMP) which functions as a message multiplexor/concentrator in much the same way as an ARPANET IMP.⁴ The MENEHUNE accepts messages from the UH central computer, an IBM System 360/65 running TSO (as of December 1974, a 370/158) or from ALOHA's own time-sharing computer, the BCC 500, or from any ARPANET computer linked to the MENEHUNE via the ALOHA TIP.⁵ Outgoing messages in the MENEHUNE are converted into packets, the packets are queued on a first-in, first-out basis, and are then broadcast to the remote users at a data rate of 9600 baud.

The packet consists of a header (32 bits) and a header parity check word (16 bits), followed by up to 80 bytes of

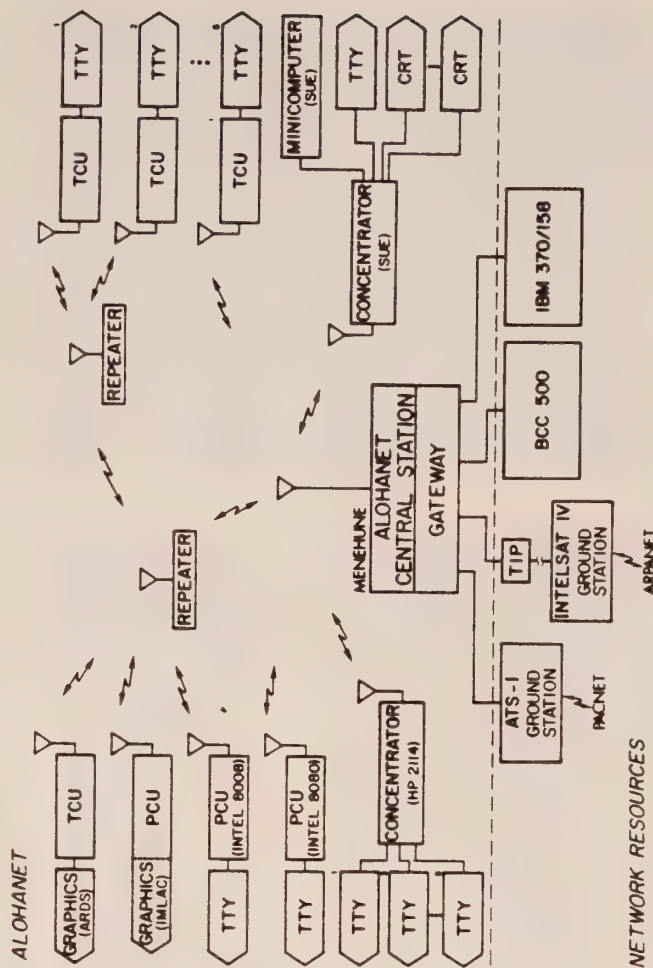


Figure 1 The ALOHANET

data and a 16-bit data parity check word. The header contains information identifying the particular user so that when the MENEHUNE broadcasts a packet, only the intended user's node will accept it. More will be said about packet formats later.

The random access channel (at 407.35 MHz) for communication between users and the MENEHUNE is designed specifically for the traffic characteristics of interactive computing. In a conventional communication system a user might be assigned a portion of the channel on either an FDMA or TDMA basis. Since it is well known that in time-sharing systems, computer and user data streams are bursty,⁶ such fixed assignments are generally wasteful of bandwidth because of the high peak-to-average data rates that characterize the traffic. The multiplexing technique that is utilized by the ALOHANET is a purely random access packet switching method that has come to be known as the pure ALOHA technique.⁷ Under a pure ALOHA mode of operation, packets are sent by the user nodes to the MENEHUNE in a completely unsynchronized manner—when a node is idle it uses none of the channel. Each full packet of 704 bits requires only 73 msec at a rate of 9600 baud to transmit (neglecting propagation time).

The random or multi-access channel can be regarded as

a resource which is shared among a large number of users in much the same way as a multiprocessor's memory is "shared". Each active user node is in contention with all other active users for the use of the MENEHUNE receiver. If two nodes transmit packets at the same time, a collision occurs and both packets are rejected. In the ALOHANET, a positive acknowledgment protocol is used for packets sent on the random-access channel. Whenever a node sends a packet it must receive an acknowledgment message (ACK) from the MENEHUNE within a certain time-out period. If the ACK is not received within this interval the node automatically retransmits the packet after a randomized delay to avoid further collisions. These collisions will limit the number of users and the amount of data which can be transmitted over the channel as loading is increased.

An analysis⁸ of the random access method of transmitting packets in a pure ALOHA channel shows that the normalized theoretical capacity of such a channel is $1/e = 0.368$. Thus the average data rate which can be supported is about one sixth the data rate which could be supported if we were able to synchronize the packets from each user in order to fill up the channel completely. Put another way, this result shows the present 9600 bit/second

channel could support between 100 and 500 active teletype users—depending upon the rate at which they generate packets and upon the packet lengths.

ALOHA/NET remote units

The original user interface developed for the system is an all-hardware unit called an ALOHA/NET Terminal Control Unit (TCU), and is the sole piece of equipment necessary to connect any terminal or minicomputer into the ALOHA channel. As such it takes the place of two dedicated modems for each user, a dial-up connection and a multiplexor port usually used for computer networks. The TCU is composed of a UHF antenna, transceiver, modem, buffer and control unit.

The buffer and control unit functions of the TCU can also be handled by a minicomputer or a microcomputer. In the present system several minicomputers have been connected in this manner in order to act as multiplexors for terminal clusters or as computing stations with network access for resource sharing. A new version of the TCU using an Intel 8080 microcomputer for buffer and control has been built. Since these programmable units allow a high degree of flexibility for packet formats and system protocols, they are referred to as PCU's (Programmable Control Unit). A more detailed discussion of terminal considerations is given in a companion paper in these proceedings.¹

Since the transmission scheme of the ALOHA/NET is by line-of-sight, the radio range of the transceivers is severely limited by the diversity of terrain (mountains, high rise buildings, heavy foliage) that exists in Hawaii. A recent development has allowed the system to expand its geographical coverage beyond the range of its central transmitting station. Because of the burst nature of the transmissions in the ALOHA channel it is possible to build a simple store-and-forward repeater which accepts a packet within a certain range of IDs and then repeats the packet on the same frequency. Each repeater performs identically and independently for packets directed either to or from the MENEHUNE. Two of the repeaters have been built which extend coverage of the ALOHA/NET from the island of Oahu to other islands in the Hawaiian chain. These repeaters are discussed in more detail in the following section.

PROTOCOL CHOICES

Two fundamental choices which have dictated much of the system protocol are the two-channel star configuration of the original network and the use of random accessing for user transmissions. Investigation of the random accessing principle using radio was in fact the original motivation for constructing the ALOHA/NET, while the two-channel configuration was primarily chosen to allow this investigation without complication from the relatively dense total traffic stream being returned to all users. An additional reason for the star configuration was the desire to

centralize as many communication functions as possible at the MENEHUNE, minimizing the cost of the TCU at each user node.

Within this context, a number of protocol issues must be resolved. The more important of these are:

- random access channel control
- broadcast channel queuing
- packet length
- addressing
- error control
- flow control

Many of the original choices in these areas have undergone significant changes as a result of new user resources and user interfaces, or in some instances due to advancements in theoretical knowledge. The addition of repeaters has (potentially) a particularly significant impact on protocol.

We now discuss some of the considerations and resulting choices made in each of the above areas, with the impacts of new factors introduced within the context of each area. The section concludes with a brief discussion of the problem of integrating file traffic into the random access channel, a subject of current concern in the ALOHA/NET.

Random access channel control

The retransmission strategy used in the random access scheme plays a central role in the scheme's effectiveness. Its determination directly affects the average delay experienced by users for a successful transmission, given a certain number of users accessing the channel, their traffic statistics, and the channel capacity. It can also be used to prevent the occurrence of channel saturation, a situation in which the channel becomes filled with retransmissions and the number of successful packets falls to zero. These topics have only recently been quantified^{2,3} and remain subjects of current investigation.

One approach is to use different constant retransmission intervals at each node, with the intervals equal to integer multiples of the maximum packet transmission time to avoid subsequent conflicts. This results in a priority structure, since nodes assigned the longer intervals will experience a correspondingly longer average delay. As the number of nodes becomes large, however, unacceptably large delays result for the majority of users.

A strategy more appropriate for large user populations is to randomize the retransmission intervals used at each node (note that a priority structure can still be introduced if desired by using larger mean values for lower priority users—in the remaining discussion, equal priorities will be assumed). According to recent results by Lam,⁴ the resulting channel behavior appears to be relatively insensitive to the exact nature of the randomization, at least when comparing the use of uniform and geometric distributions. In any event, the cost of implementing a particular distribution at each node is an important design consideration.

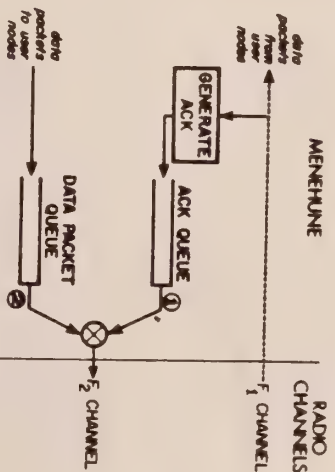


Figure 2—Broadcast Channel Multiplexing

Based on initial estimates of the expected ALOHA/NET characteristics, a choice was made to use a uniform distribution. This allowed a relatively simple implementation in both hardware and software user nodes.

A simple technique was used in the original system nodes to achieve short delays when the channel is lightly loaded, while preventing channel saturation from occurring due to peak-hour loading or statistical traffic fluctuations: small retransmission intervals are used (relative to the intervals between new packets), but only for a maximum of three successive retransmission attempts. If the third attempt is unsuccessful, the user is notified of a failure and must manually reinitiate the retransmissions. This in effect introduces a long interval between every three retransmissions, allowing time for retransmissions from other users to succeed. Based on a maximum packet transmission time of 70 milliseconds, the intervals are selected from a range of 0.2 to 1.5 seconds, giving a mean of about 0.7 seconds (ten maximum packet times) per retransmission. The lower bound is chosen to allow sufficient time to receive an ACK from the MENEHUNE if the packet was sent successfully, avoiding unnecessary retransmissions. (This time is based on a direct user-MENEHUNE path; if repeaters form a part of the radio path, the lower limit must be increased accordingly.)

The newer programmable PCU's in the system offer the capability of a more flexible strategy, for example allowing the interval used after each third retransmission to be automatically inserted. The use of different strategies, such as continuously increasing the time range used for selection of successive retransmissions, is also easily implemented by program; these and other strategies are currently under investigation.

Broadcast channel queuing

The MENEHUNE acts as a concentrator for the broadcast (F_1) channel, queuing waiting traffic when necessary for sequential transmission to user nodes. Four

complicating factors exist, however: a need for priority queuing, fair allocation of the channel, the turnaround delay required by half duplex nodes, and the presence of repeaters.

Priority queues

It is important that the F_2 channel data traffic not prevent the prompt return of an ACK to a user node, since this could lead to unnecessary user retransmissions and possible degradation of the random access (F_1) channel. Thus, an integral part of the F_2 channel multiplexing is the priority queuing mechanism maintained by the MENEHUNE, as shown in Figure 2. Whenever a transmission is completed on the F_2 channel the ACK queue is checked, and if not empty the ACK at the head of the queue is sent. Only when the ACK queue is empty is the data packet queue checked for waiting packets. This guarantees that at most one complete data packet plus any previously queued ACK's will be sent ahead of an ACK just placed on the queue. (Because the average rate of successful arrivals on the F_2 channel is limited to one-sixth the rate of F_1 transmissions by the random access technique, the number of previously queued ACK's will be zero most of the time.)

Fairness

A second problem is the possible hogging of the F_2 channel by one or a few users. This problem is eliminated by the queuing discipline used for the data packet queue. Only one packet per user is allowed on the queue at any time, and the queue is serviced on a first-come-first-served (FIFO) basis. The prevention of more than one packet per user on the queue is handled in conjunction with user flow control, discussed below.

Turnaround delay

A delay function is used by the MENEHUNE to count off the time required by half-duplex user nodes to switch from a transmit to a receive state. The actual time is determined by the equipment type—the original off-the-shelf equipment required 100 milliseconds due to its use of mechanical relays; approximately 10 milliseconds is counted off for newer equipment now in use.

Repeater scheduling

The addition of repeaters to the system introduces a number of new problems into the F_2 channel, both because of radio range overlap and the nature of the repeaters themselves. The latter are store-and-forward devices; a packet which is to be repeated is first received and stored in its entirety, then transmitted on the same frequency on which it was received (preventing reception of a new packet during this time). In order to prevent the loss of a

second packet destined to the same repeater, the MENEHUNE must therefore appropriately schedule the packets in its F_1 channel queues.

For efficient scheduling (i.e., to maximize channel utilization), the MENEHUNE must know the repeater routing path for each user node. This function could thus become quite complicated or even not achievable, depending on the degree of dynamic routing used. Because of the small percentage of traffic currently handled by repeaters in the present ALOHANET, a very simple brute force method is used: whenever a packet is sent which is forwarded by one or more repeaters, the MENEHUNE counts off sufficient time for it to be repeated once before beginning a new transmission to any node (knowledge of which packets are to be repeated is available from the user address, discussed below). This results in wasted channel capacity, but is not significant due to the capacity available in the system at present.

Packet length

Three factors having an important impact on the system are the use of variable or fixed-length packets, the way packet length or the number of data bytes is indicated, and the maximum packet length allowed. The choices made must take into account the different traffic characteristics generated by line-oriented and character-oriented user-computer interactions.

Line transmissions

Fixed-length packets were used in the initial system to simplify the design and construction of system hardware. The data packet length for both channels was chosen to allow up to 80 data bytes (640 bits), based on the user delays introduced by the 9600 bps channel data rates, the line length of the terminals in the system, and the line-oriented characteristics of the IBM 360/65 used as the central time-sharing system. An end-of-line (EOL) indicator consisting of eight zero bits was used within the packet to identify the end of actual data, where the latter was restricted to 7-bit ASCII with the eighth (parity) bit set to one. Since it was anticipated that many of the lines typed by users would be less than 40 characters, a second packet type was also defined which contained a 40-byte data field (a "Half-Packet"). This last step proved to be a significant source of both hardware and software bugs.

The packet formats have since been changed to allow the use of variable-length packets with newer user nodes. An 8-bit count field is used in the packet header to indicate the number of 8-bit data bytes in the packet, with the data parity word immediately following the last data byte. In addition to eliminating the wasted channel capacity of the fixed-length packets, this also removes constraints on the data themselves necessitated by unambiguous detection of the EOL indicator within the data stream. The 80 data-byte maximum has been retained for both channels, since

it still appears to be a reasonable upper bound with respect to both the multiplexing delays introduced to either channel and node buffering requirements. This should not be construed as an indication that this length is optimal, however, as file-oriented messages are introduced to the total traffic and/or user node storage continues to become cheaper, a larger maximum may be desirable for one or both channels (for a given channel data rate and user response time constraints).

Character-by-character

The increased flexibility provided by PCU's has allowed the introduction of a "short" data packet in which a single data byte is sent in the header in place of the byte count, followed only by the header parity word. Although a use for this packet occasionally arises for interactions with line-at-time systems, its main use is with the character-oriented ARPANET computers now available to ALOHANET users.

The use of these character-oriented systems can have a considerable impact on the size and frequency of packets sent in the random access channel. This has an important consequence for the buffering strategy and choice of packet length used at each node: since a new transmission cannot begin until an ACK has been received for the last one, all characters typed by the user during the ACK waiting time should be sent in a single packet. Thus if communication delays tend to overlap inter-character generation times, the affected characters are accumulated at the originating node and sent (more efficiently) in a variable-length packet, without adversely affecting user-computer interaction.

A logical extension of this last strategy is to buffer all characters typed by the user at his node until one is typed which causes some action to be taken by the computer. If the appropriate set of action characters is known at the user node, this allows an optimum use of both channel capacity and system buffering without degrading the user-computer interaction. A scheme which allows this to be done in conjunction with echoing control is given by Davidson,¹² and is currently being introduced into selected ARPANET hosts. Its implementation cost in ALOHANET PCU user nodes appears reasonable, and is anticipated for use as its support by host computers becomes widespread.

Addressing

User nodes

User addressing is determined by the radio channel configuration and associated multiplexing technique. Ignoring repeaters for the moment, the two-frequency configuration used in the ALOHANET allows only a single destination in the random access channel (the MENEHUNE), and a single source in the broadcast channel (the MENEHUNE). Thus only the sender's address is required in the random access channel and only the destination ad-

dress in the broadcast channel, which in both cases is the user address. Concentration of more than one user at a radio node is handled by permanently allocating a block of user addresses to the node, allowing user node multiplexing without introducing another level of addressing complexity to the system. The required address space is determined by the total number of users expected to be supported by the random access channel, and is 2^8 (eight header bits) for the present 9600 bps ALOHANET channel.

Repeaters

The use of repeaters in the system introduces some significant new factors to be considered in choosing an address scheme. Because of radio range overlap and the store-and-forward nature of the repeaters, problems can arise involving conflicts generated by two or more repeaters repeating simultaneously to the same destination, infinite repeating of the same packet (looping), and weak-signal operation due to multiple (but time-sequential) paths. In addition, the addressing scheme directly affects the MENEHUNE's ability to schedule transmissions in order to maximize broadcast channel utilization, as discussed in a preceding section. The ability to eliminate or minimize these problems depends on the degree of mobility desired for user nodes and/or the repeaters themselves.

Because of the small percentage of user nodes which currently require repeaters in the ALOHANET, a simple scheme is in use based on the hardwired properties of the original repeaters built for the system. A block of user addresses is defined for each repeater, the latter repeating only those addresses in its block. The block assigned to a repeater two hops from the MENEHUNE is a subset of the block assigned to its first hop repeater. User nodes are constrained to operate within the geographic range of their "assigned" repeater by this scheme, but the node's user address is easily changeable if a relocation becomes necessary. Since only one path choice exists between each user node and the MENEHUNE at present, the optimum path is selected by default. As the number of repeaters in use increases and existing units are replaced by programmable devices, a more flexible repeater addressing scheme is expected to be implemented.

Resource addressing

This refers to the user's choices regarding which system resource he may communicate with. The system allows users to request a connection to the campus IBM 370/158, the ARPANET, or another ALOHANET user node. This is accomplished by sending special sequences of ASCII characters in the data portion of packets to the MENEHUNE, which may either be typed by a terminal user or automatically generated. If the requested destination is available, its identification is stored in a Connec-

tion Table entry for the requesting user in the MENEHUNE, and the user's address stored in a similar entry for the destination. All subsequent packets from the user are passed to the stored destination and conversely, until either end requests that the connection be broken.

Two exceptions exist to this connection table routing of packets. The first are commands intended for the MENEHUNE, such as the "connect" and "disconnect" above. The second is a capability which allows a user to send a single packet to another ALOHANET user independently of current connection table entries. The originating user simply types a special two-character ASCII sequence followed by the destination user's address (up to three ASCII digits), followed by the desired text.

Note that in the case of a connection to another ALOHANET node, the latter's address is also the resource address. If the node's resource can service more than one user at a time (such as might be the case for a specialized minicomputer or storage device), the present addressing scheme requires either that a block of addresses be allocated to the receiving node (as in the case of a concentrator for sending), or a sub-address be sent in the text portion of every packet. The block allocation suffers from rigidity in that resource addresses cannot be reused dynamically by different users, and does not appear desirable if many such addresses must be allocated in the system.

Error control

Random-access channel

Two distinct error sources exist at the MENEHUNE receiver, the usual random noise and errors due to packet conflicts. Because of the high probability of errors due to conflicts at full loading of the random access channel, a very reliable error detection mechanism is required. To achieve this it was decided to use two 16-bit cyclic polynomial parity check words in each data packet, one following the header and a second following the data. The separate header parity check forms the basis for a highly reliable packet synchronization method discussed in another part of this paper; it also allows reliable establishment of packet length and other information prior to processing the data portion of a packet. A single header bit is also used in conjunction with the parity check for sequence numbering, allowing the detection of duplicate packets by the MENEHUNE.

Broadcast channel

Error control for broadcast channel data packets (MENEHUNE to user nodes) involves some special considerations. For efficient operation, the usual positive acknowledgment scheme in which the ACK's themselves are not acknowledged depends on a high probability of the ACK's being successfully received. However, an ACK sent

from user nodes must compete with data traffic in the random access channel. At full channel loading each random access packet must be retransmitted an average of 1.7 times, which means each data packet or ACK must be sent a total of 2.7 times on the average before it is successfully received*. But in order to force retransmission of the ACK's, the data packet being acknowledged must also be sent an average of 2.7 times by the MENEHUNE, even though it was received correctly the first time! The problem is compounded by the typically high ratios of computer/user traffic which exist for most interactive systems, resulting in many more ACK's than data packets in the random access channel. This problem was "resolved" for the initial implementation by simply not sending ACK's from user nodes. Because of the high receiver signal strengths at the nodes, a very low error rate was anticipated; considering also that user nodes consisted only of human terminal users, it was decided that a simple error detection/user notification scheme would be sufficient.

However, this is in general not adequate when more sophisticated data transfer functions take place or significant error rates exist at user nodes. An example of the first case is the loading of programs into core storage of a mini-computer node, where manually initiated error recovery usually requires restarting the loading from the beginning of the file. In the second case, error rates can become appreciable when user nodes are located in weak signal areas caused by distance, multipath interference, or line-of-sight blocking, or in strong signal areas in which strong local noise sources also exist. To allow for these situations, an option which allows user nodes to send positive acknowledgments has been implemented. The scheme works identically to that for the random access channel, but is only used selectively with newer programmable nodes when required (it can be turned on or off by a command from the user node to the MENEHUNE). Its effectiveness is based on the relatively light existing channel loading of the system and its use by only a few of the nodes.

One solution to this problem when all traffic to user nodes must be acknowledged in a loaded random access channel is to use sequence numbering with a large modulus, sending an ACK only when the maximum sequence number is received. This approach suffers from the unpredictable nature of interactive user-computer traffic; however, if the last computer output prior to new user input is missed by the node, a potential deadlock situation is created until the user decides something is wrong and takes manual action. An additional mechanism can be used to circumvent this, such as using automatic timeouts at the user node or sending dummy traffic to the node to "flush out" missed packets. However, the sequence numbers succeed only in reducing the number of ACK's sent in the random access channel—to eliminate the unnecessary

repetitions of data packets from the MENEHUNE, it is also necessary to acknowledge the ACK. That is, the ACK sent by a user node is timed out and retransmitted until an acknowledgment for it is received, just as for data packets. If another packet is waiting for transmission to the node at this time, its transmission with the next sequence number constitutes the ACK to the ACK; otherwise, a short ACK-ACK packet is sent by the MENEHUNE. This can be easily shown to result in significantly less total channel overhead, at the expense of more complication in the node implementation.

Repeaters

We have so far ignored the effects of repeaters in this discussion on both random access and broadcast channel error control. The repeaters currently in use in the ALOHNET do not generate acknowledgments in either direction, resulting in only end-to-end acknowledgments between the MENEHUNE and user nodes as above (but with longer minimum retransmission timeouts). This choice was made for initial repeater simplicity; it has been shown analytically, however, that a hop-by-hop acknowledgment scheme is in general superior to an end-to-end scheme, at least in contexts such as ARPANET and the ARPA Packet Radio effort.¹² Thus we expect to convert to a hop-by-hop scheme when the existing repeaters are replaced by programmable units and/or repeater traffic error rates require it; this area remains a relatively unexplored problem domain within the present ALOHNET implementation.

Single-channel configurations

Finally, we note that the problems discussed above concerning ACK's sent by user nodes in the random access channel are effectively non-existent if a single-frequency channel configuration is used (and propagation times are less than the shortest packet transmission times). If all nodes can hear the transmission of all other nodes, it is only necessary that nodes refrain from sending for an ACK packet time following the transmission of a data packet by any node, except for the intended receiver who sends an ACK (if appropriate) during this time. Thus ACK's are sent conflict-free, allowing a simple positive acknowledgment scheme to be used for all traffic. Note that packets sent by the MENEHUNE are treated exactly the same as packets sent by user nodes with respect to ACK's, thus also eliminating any effects due to asymmetric computer-user traffic ratios.

Flow control

The initial system

In the initial system environment of a single half-duplex time-sharing system, model 33 Teletypes, and hardware

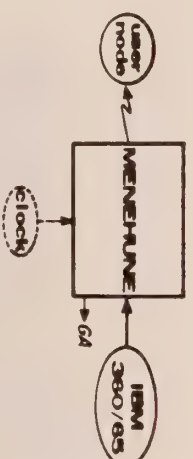


Figure 3. Broadcast Channel Flow Control (Original System)

user nodes which buffered only the line being displayed, flow control was a relatively simple matter. A user always received at least one output line from the time-sharing system (IBM's TSO running on a 360/65) for each input line, and a prompt character when it was ready for more input. The bandwidth between the MENEHUNE and 360 and the latter's I/O response times are such that one or two MENEHUNE buffers are normally sufficient to support transfers of packets received from the random access channel; in the unlikely event that no buffers are available when a packet arrives, the channel protocol guarantees its retransmission. Thus no explicit flow control was provided to prevent new packets from being sent by a user node. If the user sends one before the 360 is ready, the packet is discarded and a "WAIT" message returned to the user by the MENEHUNE (the status of each 360 connection is known in the MENEHUNE by information routinely passed from the 360).

Broadcast channel flow control was necessary, however, since each line (packet) sent to a (hardwired) user node must be completely displayed before a new line can be received. This was accomplished by the scheme shown in Figure 3, in which the control for each user node is centralized at the MENEHUNE. The latter counts off the required display time following transmission of each packet to a user, inhibiting further transmissions to that user until the time is up. To prevent 360 output from tying up MENEHUNE buffers while packets are being displayed, a handshaking flow control is used: the 360 sends only one line of output for each user, then waits for a go-ahead (GA) message with that user's address. The GA time is up, resulting in at most one buffer required for each user (the MENEHUNE can also hold up acceptance of any packet from the 360 indefinitely until it has buffer space available). Note that this strategy also prevents any user from hogging the broadcast channel, since it allows only one packet per user in the channel queue.

Some terminal complications

The introduction of high speed CRT and hardcopy terminals to the system required an expansion of the MENEHUNE's flow control mechanism for the broadcast channel. A set of display rates was added, with the rate used at each user node stored in a permanent table in the

MENEHUNE; a user can change the stored value for his node by typing a special command to the MENEHUNE at any time. The CRT terminals require an additional flow control mechanism to suspend output when the CRT screen has filled, allowing the user to signal when he is ready to proceed. Thus a screensize command was created which allows users to specify a screensize of between one and 99 lines (for an infinite screensize); this value is also stored in MENEHUNE tables for each user node. A counter is maintained for each user with a finite screensize specification and is updated for each line sent to the terminal; when the maximum is reached, the MENEHUNE suspends generation of the GA message until the user inputs a carriage return.

Satellite complications

The next complication to MENEHUNE flow control processing was caused by the connection of the ALOHNET to the ARPANET. The latter involves a 50 Kbps INTELSAT IV satellite path connecting Hawaii to California; because of its long propagation time (approximately 0.25 seconds) and ARPANET flow control protocol, a large amount of buffering is required at the receive end of the link to support continuous display at higher speed terminals—in particular, a 9600 bps terminal requires approximately a 1000-byte buffer. (Since in general CRT terminal users do not require continuous output at this rate, a smaller amount of buffering is in fact used.) This required a substantial increase in the size of the MENEHUNE buffer pool and a more complicated queuing structure to support the broadcast channel, since now more than one packet per user must, in general, be stored in the MENEHUNE during display at the user node. To maintain the single-packet-per-user policy for the channel queue, a separate queue was created for each user to hold additional packets. The resulting flow control scheme is shown in Figure 4, where the GA's sent to the 360 in Figure 3 are now sent to the internal ARPANET protocol module. The maximum allowed size of each user queue is determined by the user's terminal rate and the available MENEHUNE buffer pool, and in turn defines

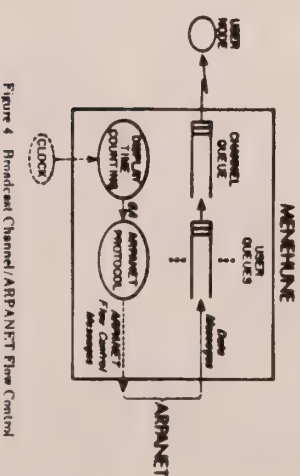


Figure 4. Broadcast Channel/ARPANET Flow Control

*This assumes ACK's and data packets are the same length, although the ACK's are in fact shorter; the resulting error rate is still very high compared to a typical conflict-free channel.

File traffic

the parameters used in the ARPANET flow control protocol.

Multiple-line packets

A second complication resulting from the ARPANET connection concerns the extra time required by some higher speed displays for certain characters such as carriage return (CR) and/or line feed (LF). Output from the 360 in the initial system contained such characters only at the end of a line (packet), allowing the transmission time and other inter-packet delays to provide any extra time required. However, many ARPANET computers are character-oriented, at times generating many CR and LF characters within a single packet. Thus it was necessary to provide a padding function in the MENEHUNE which inserts dummy characters or otherwise adds a display time delay after each CR or LF occurrence within packets destined for a higher speed (greater than 110 bps) terminal. This necessitates the splitting of packets whenever the maximum 80-byte packet length is exceeded, and in general involves a significant amount of additional processing per packet.

Full duplex interaction

A third complication arising from many ARPANET computers is their full duplex user interaction. Unlike the 360, users do not necessarily receive output in response to each input or an indication of when the computer is waiting for more input. Since no explicit flow control is provided for input from user nodes to the MENEHUNE, users are forced to either interact in a half duplex fashion (guessing as to when the computer has finished its output) or suffer occasional losses of input data and subsequent retyping. The latter can occur frequently with the hardwired TCU's, since they contain a single buffer which is used for both keyboard input and display; if computer output arrives while the user is typing, the typed characters are overwritten in the buffer by the received packet. The newer programmable user nodes now in the system provide full duplex buffering for the terminal, allowing a packet to be received and displayed without disturbing the keyboard buffer.

However, even if user nodes are completely full duplex a flow control problem exists for packets sent to the MENEHUNE. Unlike the case for the 360, users of full duplex hosts may generate successive input packets without receiving responses from the host computer. If the ARPANET or host computer or both slow down, an excessive number of buffers can become queued in the MENEHUNE on behalf of the user. Thus, to prevent user hogging of the buffer pool a count of the number of input buffers queued for each user is now maintained, when equal to the maximum allowed, arriving packets are discarded and a discard notification returned to the user.

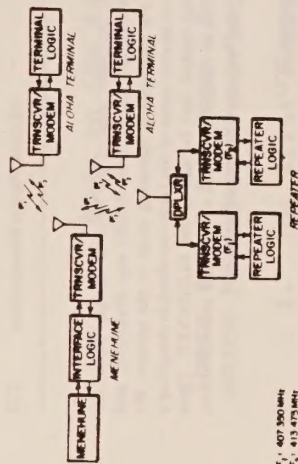


Figure 5—ALPHA System UHF Radio Communication System

sources, regulatory requirements, and equipment costs. In the case of the ALOHANET, a wide channel bandwidth was considered desirable for the random access channel since user nodes are required to send messages to the MENEHUNE at high peak data rates compared to their average data rate. Wide bandwidth was also deemed advisable for the broadcast channel due to the expected high traffic density from the MENEHUNE. The use of wide channel bandwidth tends to force the use of higher frequencies where spectrum crowding is less severe and the availability of bandwidth is greater. Crowded radio bands are undesirable not only from the standpoint of interference to other users but also because of potential interference from them. Another disadvantage of lower frequencies is the higher probability of interference from man-made noise sources, particularly in an urban area where the ALOHANET has most of its terminals.

From the above considerations it can be seen that the system's communication requirements tend to emphasize the use of higher radio frequencies. The primary constraint on moving to even higher frequencies is equipment cost and radio range. Above 500 MHz equipment costs tend to escalate rapidly. Area coverage also becomes more difficult due to more pronounced shadowing effects of the radio waves by buildings and hilly terrain. (Above 30 MHz radio propagation tends to be limited to line-of-sight paths.)

Therefore, the 400 to 500 MHz UHF band was selected as the optimum for the ALOHANET radio frequencies. Reasonably priced commercial radio equipment was found to be available in this frequency region and radio band crowding was not severe in Hawaii. Initially, assignments in the 450 to 470 MHz mobile radio band were requested but were rejected by the FCC because of our wide channel bandwidth requirements. (The mobile radio channels are specified at about 15 KHz bandwidth, whereas we were requesting 100 KHz.) We were fortunate enough to receive assignments as an experimental service in the government UHF band of 406 to 420 MHz, where spectrum space was available.

Since most radio equipments available in the UHF bands use frequency modulation (FM), this type of

modulation was selected for the RF channels. A slight variation was incorporated in the hardware design to minimize the interface problems between the radios and the data modems. This variation was the use of a subcarrier tone to carry the actual data modulation. This tone is phase-shift-keyed by the data and the resultant signal is used to modulate the FM transmitter. This modulated tone is recovered from the FM receiver and fed to the demodulator of the modem. This modulation system is referred to as FM/DPSK to indicate frequency modulation by a differentially phase-shift-keyed subcarrier. (Differential phase-shift-keying is used to resolve the problem of received phase ambiguity.) The resultant configuration is shown in Figure 5.

Radio range

The maximum operating distance between any terminal of the ALOHANET and the MENEHUNE (or a repeater) is specified as the system's radio range. This distance is primarily a function of a transmitter's radiated power, the receiver's sensitivity, and the attenuation of radio signal power for the given distance. Local noise conditions at the receiver location can also affect this distance, but for system planning purposes, range is usually calculated on the basis of some given propagation model. For line-of-sight paths, which exist at VHF, UHF, and higher frequencies, two different models are used depending upon local topographical conditions. In an urban area these paths are partially obstructed and suffer from multipath effects. A power loss proportional to $1/R^4$ is usually assumed for these conditions.¹⁴ Where paths are unobstructed and well clear of the local terrain, a spreading loss proportional to $1/R^2$ can be assumed. Receiver threshold sensitivity in the ALOHANET is defined as that receiver input power level which causes an average bit error rate of 10^{-4} . This bit error rate should provide a packet throughput reliability better than 99 percent for full-length ALOHA packets.

Assuming a transmitter equivalent radiated power of 10 watts, a simple whip antenna at a user terminal, an elevated antenna at the MENEHUNE or repeater and a 3 microvolt receiver sensitivity, the radio range works out to about 17 miles in the urban area for the ALOHANET frequencies. Between repeaters and the MENEHUNE terminals, which have well-elevated antennae and good path clearances, the assumed $1/R^2$ model gives a maximum range of 290 miles. The use of high-gain omnidirectional antenna arrays at repeater sites extends these ranges. Tests conducted on a 100 mile path between two ALOHANET repeaters confirmed the $1/R^2$ spreading loss assumption and indicated a fade margin of 30 db existed (due to the 10 db gain antennae used for the test).

Data synchronization

Because of the burst nature of radio transmission of ALOHANET packets, special synchronization techniques

must be employed in the modem and data terminal equipment. Since the phase-shift-keying used in the ALOHNET modem design is a bit-synchronous technique, bit synchronization must first be performed in the demodulator before packet synchronization can be attempted. Bit-sync is performed by a phase-locked circuit, and a lock-indication signal is passed to the data equipment when bit-sync has been attained. The bit-sync detection circuit is so designed to provide a very low false detection probability (less than 10^{-7}) and a high probability of packet detection. The narrow bandwidth of the phase-locked circuit presently designed into the ALOHNET modem requires a bit-sync preamble of 90 bits to ensure reliable bit-sync. Studies have indicated that this preamble can be reduced to about 10 bits by use of a redesigned wide-band phase-locked circuit. In fact, we are presently contemplating doing away with the bit-sync preamble entirely, further reducing packet overhead. The unique characteristics of the ALOHA modem design make such an approach feasible.

Packet synchronization is accomplished in the ALOHNET data terminal buffer by means of the 16-bit parity word contained in the packet header. When the parity check routine accepts the header, the packet is assumed to be synchronized. Since the parity check routine is initiated by the first bit of the header, packets can be misused due to detection of an early error bit before the header. This misprobability is presently controlled by the modem at about 10^{-7} or less, providing a packet detection probability of 99.9 percent or better. The false detection probability of this circuit is $\sim 1.5 \times 10^{-11}$, which is independent of that of the modem. Thus, the overall probability of false detection is less than 1.5×10^{-11} . Therefore, less than one out of a thousand packets will be lost due to packet sync errors and packet sync false alarms occur with extreme rarity.

USER INTERFACE CHOICES

The development of the ALOHNET user interface has been an evolutionary process, as is typical of most research developments. Since there were expected to be many user modes (as compared to the single MENEHUNE model), the primary design goals were initially set as simplicity of design and low cost. This led to the design of a hardware control unit with limited data storage capability coupled to a modem and radio transceiver. This initial design was termed a Terminal Control Unit (TCU). As experience developed with operation of the net, other functions became evident as being desirable in a TCU. At about this time the first microprocessor chips and low-cost semiconductor memory chips were becoming available in the marketplace. It was decided that a new TCU design should be initiated using these new devices since much greater flexibility and additional functions could be readily incorporated in a unit having a capability of being programmed. It was also noted that the cost of these new devices was such that a unit could be built for the same

cost or less than that of the original design. Thus, the Programmable Control Unit (PCU) was developed, and there are now several operating units in the system. We will now discuss some of the issues involved in designing a terminal control unit for use on the ALOHNET. These issues lie in the general areas of interface considerations, and the technology of microprocessors.

The original TCU

The ALOHNET was originally envisioned as a terminal network, with the TCU's interfacing human users to a half duplex, line-oriented time-sharing system. At the time of the first TCU design effort memory was relatively expensive, so in order to minimize cost a single buffer was chosen for use with both the terminal keyboard and display. (As noted earlier in this paper, when full duplex computer interactions were available in the system the single buffer was found to be quite a disadvantage.) The buffer was designed for a full line length of 80 characters, which allowed handling of both the 40 and 80 character fixed-length packets defined for the system.

Additional basic functions performed by the TCU's were generation of a cyclic-parity-check code vector and decoding of received parity code words for error-detection purposes, and generation of packet retransmissions using a simple random interval generator. If an acknowledgment was not received from the computer after the prescribed number of retransmissions, a flashing light was used as an indicator to the human user. Since the TCU's did not send acknowledgments to the MENEHUNE, a steady warning light was displayed to the human user when an error was detected in a received packet. Thus it can be seen that considerable simplification was incorporated into the initial design of the TCU, making use of the fact that it was interfacing a human user into the network.

Other functions hardwired into the TCU were the obvious requirements of checking for and generating its address, packet sequence numbering, checking to see if a received packet is an ACK packet or a data packet, and generating and checking for half- or full-packet conditions. (The control bits for these functions all reside in the header portion of the packet.)

The final consideration was the choice of standard interface signals between the TCU and the user's equipment. This was a relatively simple choice, since most equipment is designed to meet the EIA standard RS-232C interface specification. Therefore, the TCU was designed to meet this standard, which allows direct connection of most terminals in use today.

Minicomputer modes

As the ALOHNET developed, some minicomputers were interfaced into the network as concentrators for a number of terminals. Many of the logical functions performed in a TCU were now incorporated into the minis' software, with error detection and parity word

generation performed in a special hardware interface unit imposed between the minicomputer and an ALOHA modem. (This unit was very much like the encoder/decoder unit used at the MENEHUNE to interface that minicomputer to the channel.) Parallel-to-serial and serial-to-parallel conversion was also performed in this interface unit.

However, a minicomputer is an expensive device to use for these simple functions, and it requires considerable amounts of power and space. If it already exists for the purpose of performing various user-oriented tasks, then it is cost-effective to incorporate the software interface and a minimal amount of hardware for use on the ALOHNET.

The advent of the microprocessor chip changed all this. The relatively low-cost processing power demonstrated by these units made it apparent that many system options we had previously considered and discarded because of hardware complexity and cost limitations in the TCU, were now viable in a PCU. Some of these options—file transfer, remote user ACKs, single frequency operation, character-by-character transmission—were discussed in previous sections. This trend toward programmable and more powerful TCU's has thus led to the development of the ALOHA PCU, using a microprocessor to handle the TCU buffering and control functions, in addition to more complex and sophisticated functions.

Microprocessor technology

The development from the hardwired TCU concept to the fully-programmable PCU has closely followed the rapidly changing technology of microprocessors. The availability of lower-cost semiconductor memory has allowed the evolution from half-duplex to full-duplex operation in the PCU, with the beneficial side-effect of decreased logical complexity due to separation of the input and output functions. However, the first PCU developed had a hardware complexity level comparable to the TCU due to the relatively primitive structure of early microprocessor designs. This first PCU, designed with the Intel 8008 CPU, required a considerable amount of circuitry for buffering and multiplexing functions needed with this early microprocessor chip. Because of the slow speed of the chip, bit-by-bit processing was not possible and additional buffering was also necessary. But, much greater flexibility was introduced into the scope of functions which could be performed, due to its programmability.

Later microprocessor designs, such as the Intel 8080 and National IMP-16, have introduced much greater sophistication into the processor chips accompanied by significant processing speed improvements. A newer PCU design, incorporating an Intel 8080 chip, has demonstrated a considerable reduction in hardware complexity accompanied by an even greater degree of processing flexibility. For example, parity generation and checking are done in software with this prototype design.

Buffering has progressed from the simple shift-register storage devices of the TCU to the use of semiconductor

RAM devices used in the microprocessor's random-access memory. All of the micro-instructions for the Intel 8080 microprocessor PCU design reside on four PROM chips, providing 1024 bytes of microcode. The random-access memory consists of 2048 bytes of RAM.

Recent product introductions such as Intel's 3000 series bi-polar chips promise even greater reductions in chip counts and increases in processing power and speed. With machines such as these, bit-by-bit processing can be readily incorporated into software, thus further eliminating the need for external interfacing hardware and simultaneously providing greater flexibility in the implementation of additional functions. A more detailed discussion of communications microprocessors is given in a companion paper in these proceedings.³

Size and power

In the earlier versions of the TCU smaller size and power drain of the unit were not considered major design objectives. The first units were designed for ease of access and hardware modifications to these TCU's were made on a fairly casual basis. As more and more of the ALOHNET came into use, however, small size, portability and lower power drain became desirable.

Of particular interest is the possibility of designing low power battery operated portable PCU's for mobile units in the ALOHNET. Since the transmitter power need only be on for a short burst corresponding to the period of the data burst, the average power of the transmitter can be a small percentage of the peak power. Since low power and small size were not original design objectives, it appears that the construction of low power portable PCU's will involve redesign of several subsections of the PCU and some new design efforts. Of particular importance is selection of a microprocessor unit which provides a minimum power-drain computer architecture consistent with functional requirements. The modem should be redesigned to use MOS devices to minimize power drain, and the transceiver designed for minimum complexity.

CONCLUSIONS

As the system has been modified during the past several years it has become apparent that packet broadcasting architecture is remarkably flexible in its tolerance of hardware, system and protocol modifications. This flexibility follows from the packet verification algorithms which lie at the basis of packet broadcasting. The only packets accepted by a remote unit or by the MENEHUNE are packets which meet all the tests expected by the potential acceptor, and the only system resource consumed by an unaccepted packet is the capacity of the channel during the short burst of the packet duration. Thus it is perfectly feasible in a packet broadcasting network to introduce a new form of packet (new in format, new in packet length, or even new in

Bugs in the Point to Point software

There are a few bugs in the point-to-point software which have been reported and are being worked on:

1. The first message after connection has been established will contain the call sign and the CTRL-X: VE3DVV(CTRL-X)Hi John... This is because the TIP must save all data entered up to the CTRL-X in case it is a general transmission. The buffer pointer is not reset after the connection is established.
2. When you type enough characters to the TIP to fill the available buffer space, and do not give a terminator (LF), the controller hands. This is because the buffer is checked for FULL before the input character is examined to see if it is a terminator. If the buffer is full, the characters input are just ignored.
3. If the TIP backs up and the LIP buffer becomes full, RNR is sent back over the link, and the other station just keeps trying. It appears that in one or both of the stations, the N(R) or N(S) counters are messed up and the messages sent after the RR are rejected because they are out of sequence.
4. A zero length packet (no I field) (not possible to send with the distribution TIP) causes one or both stations to disconnect.

Further information will be published as it becomes available.

This raises a related issue: the desirability of distributing presently centralized protocol functions such as flow control among the user nodes. Since we have just begun to gain experience with PCU's in a packet broadcast network, we must leave this as an open question.

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modulation technique) without disturbing any unit operating with the existing scheme. Only the units designed to look for the new packets will accept these packets and all other units will simply discard them.

We plan to employ this property of packet switched channels to switch the polynomial used for error control in the present packet format. The new polynomial is available in a single IC chip and will allow the possibility of error correction as well as error detection in some cases. As remote units with new packet formats are put into operation we can continue to operate the existing remote units without modification as long as we have a single unit capable of accepting the new packet format at the MENEHUNE. As a side benefit of the introduction of this modification we also note that we have effectively doubled the number of user addresses in the system. An address in use with the old packet format may be reused with the new, since each is effectively invisible to the other.

Another result of our ALOHANE experience, current technology, and recent theoretical work on ALOHA channels, is that a single-channel network configuration appears preferable to the two channels used in our present system. The major reason why this is so has to do with the broadcast property of the single-channel system, in which all nodes can (for a given geographic range) hear the transmission of all other nodes in the net.

A number of desirable properties result from this broadcast feature. First, each node can determine if the channel is free before transmitting, greatly reducing the number of packet conflicts—Kleinrock and Tobagi" have shown analytically that this can increase the throughput of a random access channel by a factor of three to five for reasonable user delays, depending on the propagation times between nodes. Second, the problem of sending acknowledgments from user nodes is resolved in a simple manner. Third, system bandwidth can be optimally allocated to both directions of traffic by simple time-sharing of the channel. Fourth, single channel repeaters require only half the radio hardware of two-channel repeaters, and, in fact, the radio transceivers at all nodes need be only half duplex. Finally, a single-channel system constitutes a fully-connected network allowing direct communication between all nodes. A star configuration can still be imposed by protocol to direct all user traffic through a central node, but is no longer required.

It is important to note that many of the above properties are made feasible by the availability of PCU's at a reasonable cost through microcomputer technology.

